XP0100(P) IP PHONE
USER MANUAL

About This Guide

Thank you for choosing this Enterprise IP Phone which is especially designed for power users in the office environment. It features fashion and sleek design, abundant telephony applications, broad interoperability with the popular 3rd party VoIP products, fulfilling the VoIP deployment needs from enterprise and ITSP.

In this User Guide, you will find everything you need to quickly use your new phone. Be sure to verify with your system administrator that your network is prepared for configuring your IP phone. As well, be sure to read the Packing List section in this guide before you set up and use the phone.

Declaration of Conformity

Hereby, it's declared that this phone is in conformity with the essential requirements and other relevant provisions of the CE, FCC.

CE Mark Warning

This is a class B device, in a domestic environment; this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning

To avoid the potential effects on the environment and human health as a result of the presence of hazardous substances in electrical and electronic equipment, end users of electrical and electronic equipment should understand the meaning of the crossed-out wheeled bin symbol. Do not dispose of WEEE as unsorted municipal waste and have to collect such WEEE separately.

This page is intentionally left blank.

Table of Contents

Getting Started 1

Packing List 1
Assembling the Phone 2
Configuration and Registration 5
Configuring via Web Page 5
Configuring via keypad 6

Overview 7

Keypad Instruction 7
LED Instruction 8
Icon Instruction 9
User Interface 10

Customizing Your Phone 13

General Settings 13
Phone Status 13
Language 13
Time and Date 14
Keypad Lock 14
Audio Settings 15
Volume 15
Ring Tones 16
Codec Selection 17
Contact Management 18
Local Directory 18
Blacklist 19
Other Settings 19
Key as Send 19
Hot Line 20
Headset Prior 20
Dual Headset 21
Anonymous Call 21
Auto Redial 22
Auto Answer 22
Missed Call Log 23
Logo Customization 23
Programmable Keys 23
Live Dialpad 23
Replace Rule 24

Dial Now 24
Area Code 24
Block Out 25
Feature Synchronisation 26
Push XML 26
WatchDog 26

Using the Basic Call Functions 27

Making a call 27
Call Devices 27
Call Methods 27
Password Dial 28
Call Completion 29
Answering a Call 29
Answering an Incoming Call 29
Denying an Incoming Call 29
DND 29
Call Forward 30
Intercom 30
During an Active Call 31
Mute 31
Call Hold 31
Call Waiting 32
Call Transfer 32
3-way Conference 33
Network Conference 33
Voice Mail 33

Using the Advanced Phone Functions 35

Account Setting 35
TLS 35
DNS-SRV 36
Network Setting 36
PC Port Setting 36
VLAN Setting 37
LLDP 37
HTTPS 38
Maintenance Tasks 38
Administrator Mode 38
Administrator/User Password 39
Reboot 39
Reset to Factory 40
Export/Import Configuration File 40
Firmware Update 40

Decryption 41
Zero-sp-touch 42
System Log Export 42
PCAP Trace Export 42
802.1X 43
DSS Keys Configuration 43
Line 44
Speed Dial 44
BLF 45
BLF List 45
Voice Mail 46
Pick Up 46
Group Pick Up 46
Call Park 47
Intercom 47
DTMF 47
Prefix 48
Local Group 48
Conference 48
Forward 49
Transfer 49
Hold 49
DND 50
Redial 50
Call Return 50
Record 50
URL Record 51
Multicast Paging 51
Group Listening 52
Public Hold 53
Private Hold 53
Shared Line 53
Automatic Call Distribution (ACD) 54
Zero-sp-touch 54
URL 55
SNMP 55
Action URL 56
Action URI 57
Music on Hold 58
SIP Diversion Display 59
Suppress DTMF Display 59
Tone Settings 60
Voice 60
Ring 61

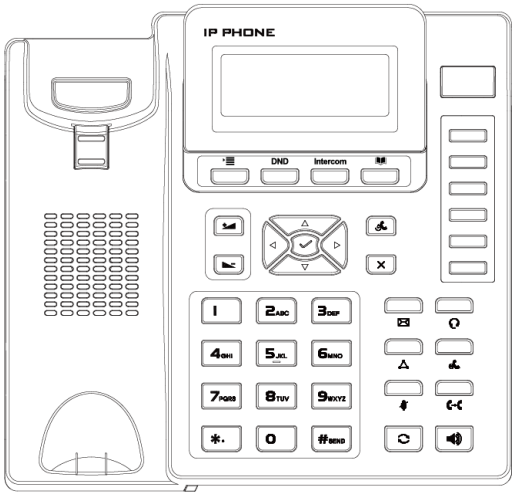
Trouble Shooting 63

Getting Started

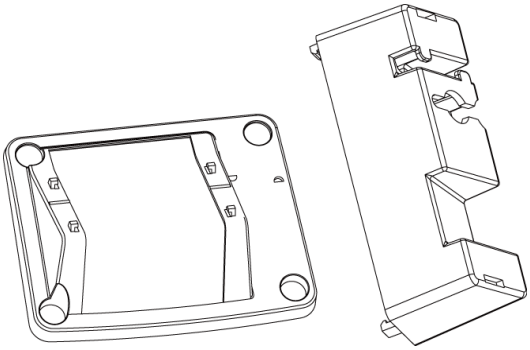
Packing List

The following components are included in your package:

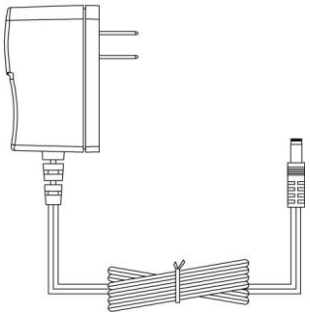
- Enterprise IP Phone



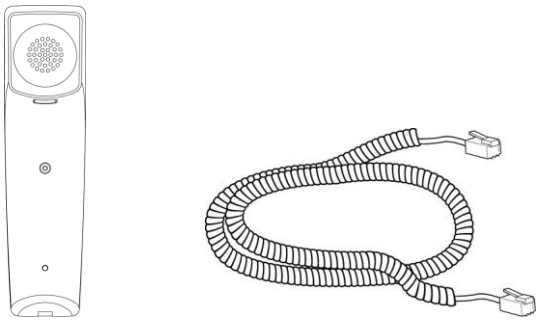
- Phone Stand



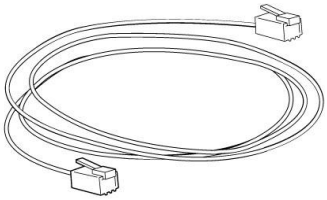
- Power Adapter



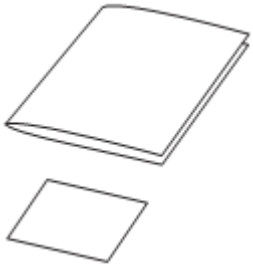
- Handset & Handset Cord



- Ethernet Cable



- Quick Installation & Quick Reference



- CD-ROM (Optional)

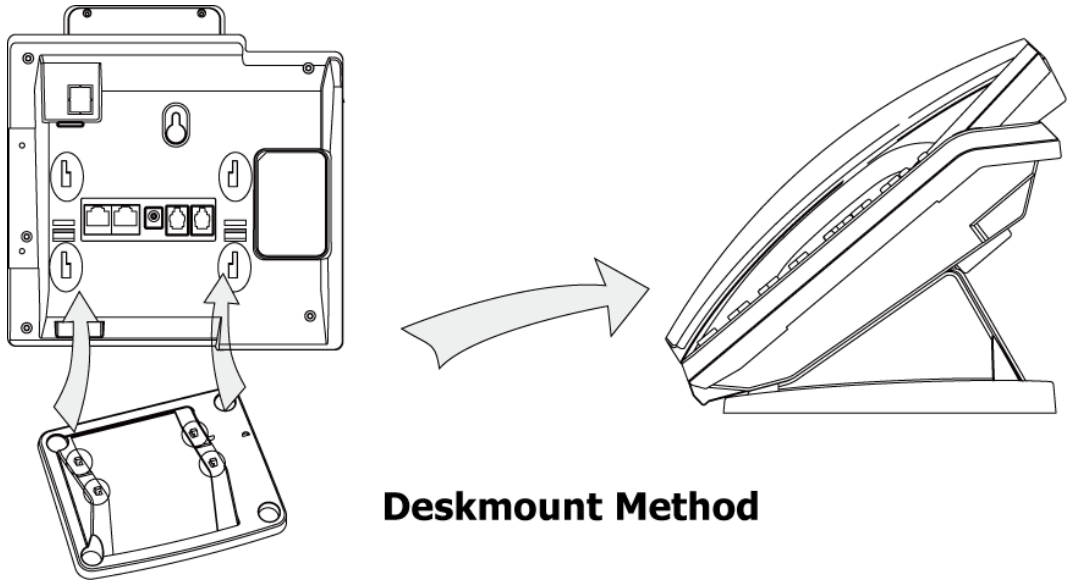
Check this list before installation to ensure that you have received each item. If you are missing any items, contact your IP phone reseller.

Assembling the Phone

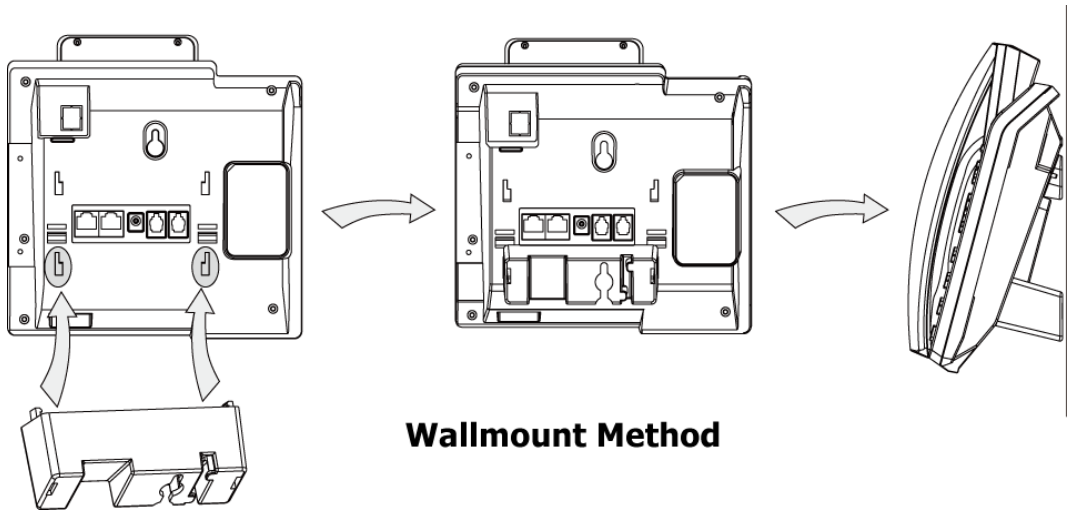
This section introduce how to assemble the phone with the components in the packing list:

- Attach the Stand
- Connect Handset and Headset
- Connect Network and Power

1) Attach the Stand, as shown below:

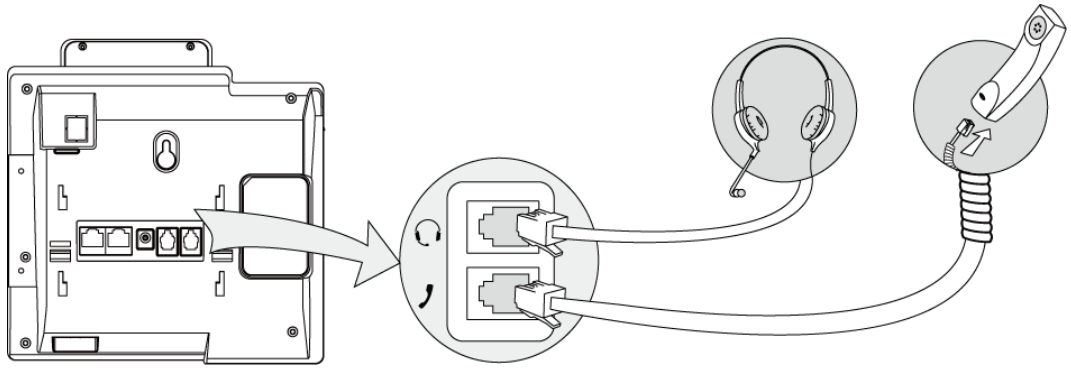


Deskmount Method



Wallmount Method

2) Connect Handset and Headset, as shown below:



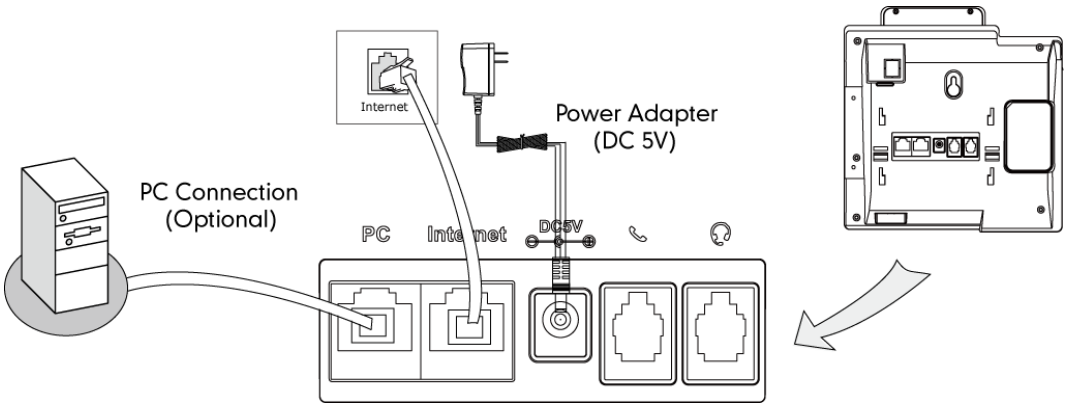
Note:
Headset is not provided in the packing list. Please contact your distributor for more information.

3) Connect Network and Power

You have two options for power and network connections. Your system administrator will advise you on which one to use.

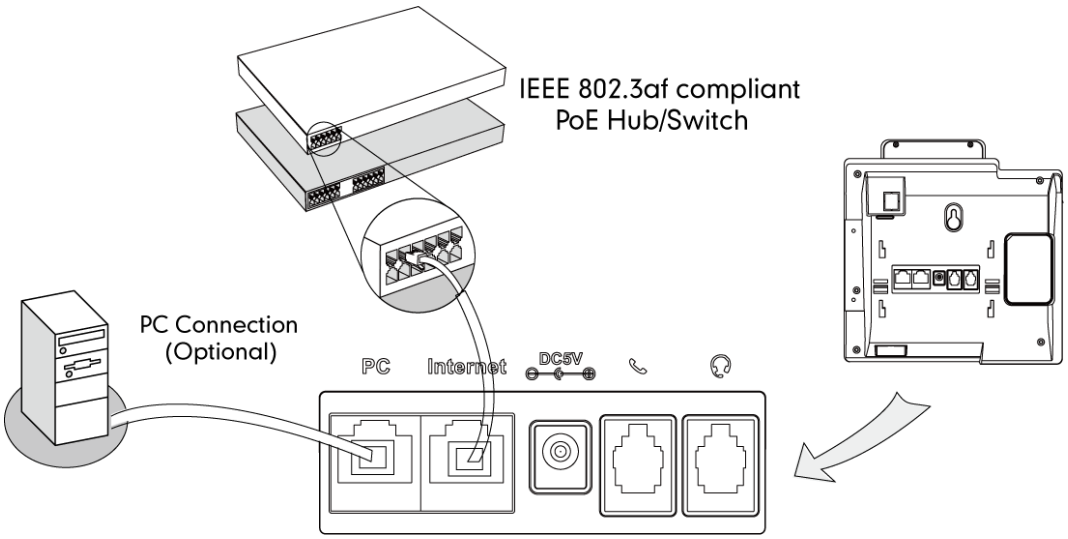
AC Power

1) Connect the DC plug on the power adapter to the DC5V port on the phone and connect the other end of the power adapter into an electrical power outlet.
2) Connect the supplied Ethernet cable between the Internet port on the phone and the Internet port in your network or switch/hub device port.



Power over Ethernet

Connect the Ethernet cable between the Internet port on the phone and an available port on the PoE (IEEE 802.3af) compliant switch/hub



Note:

1. If in-line power is provided, you don't need to connect the AC adapter. Make sure the Ethernet cable and switch/hub is PoE compliant.
2. The phone can also share the network with other network device such as a PC (personal computer). It is an optional connection.

Configuration and Registration

If you are administrator, you need to do some simple configuration to make the phone work. If not, please contact with your internet administrator or service provider for more information.

Configuring via Web Page

Press on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter the IP address (for example http://192.168.3.28) into the address bar of web browser. The default login name and password for administrator are **admin/admin**. The default login name and password for user are **user/user**.

Note:
Please locate your PC in the same network segment of IP phone (192.168.3.X) to access the web configuration page. Contact your system administrator for more information.

Network Settings
Click on Network-> **Basic**->**WAN**.

DHCP: By default the phone attempts to contact a DHCP Server in your network in order to obtain its valid network settings, e.g. IP address, sub mask, gateway, DNS server, etc.
Static IP Address: If your phone cannot contact a DHCP Server for any reason, you need to enter the network settings manually via Static IP Address. Please contact your internet administrator for more information.
PPPoE: If you are using the xDSL Modem, you can connect your phone to the internet via PPPoE mode. Please contact your ITSP for the User Name and Password for internet access.

Note:
Using the wrong network parameters may result in inaccessibility of your phone and may also have an impact on your network performance. Please contact your network administrator.

Account Settings

The phone attempts to register to the SIP server using the account/registrar data provided by the automatic or manual initialization.

Click on **Account**->**Basic**, you will find the following parameters:

Field	Description
Register Status	It shows the register status of the phone.
Account Active	You can select Enabled/Disabled to set the account respectively
Label	It is showed on the LCD for users to identify the account.
Display Name	It is showed as Caller ID when making a phone call.
Register Name	It is authenticated ID for authentication, provided by ITSP.
User Name	User account information, provided by VoIP service provider.
Password	It is provided by ITSP for registration.
SIP Server	Server for registration, provided by ITSP.

When all accounts register fail, phone will display "No Service" on the idle screen. When the phone reboot, it will register automatically. If many phones register at the same time, this will affect the server, users can set the register power up time so that the phone will random register automatically within the set time.

To configure the power up time via web user interface:

1) Click on **Network**->**Advanced**.
2) Scroll to **Registration random** field, and then enter the time in the **Registration random** field.
3) Click **Confirm** to save the change.

Note:
Should the IP PBX (SIP registrar) require an authentication, you will be prompted to enter the correct password. Make sure you are using the appropriate input method or enter the password via web user interface.

Configuring via keypad

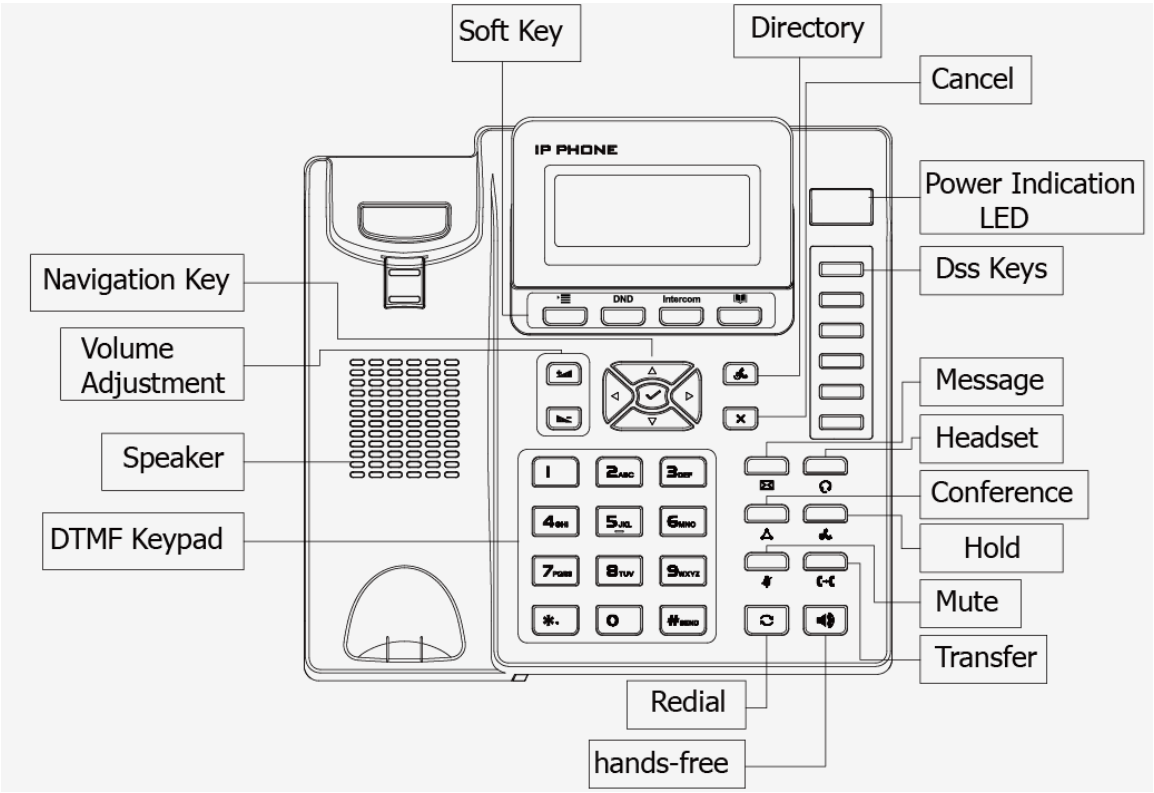
Network Settings: Press ▸≣, and then press **Settings**-> **Advanced**-> **Network**-> **WAN Port**/**PC Port** /**VLAN**/**Web Type**/**802.1x Settings** to configure the network settings.

Account Settings: Press ▸≣ , and then press **Settings**->**Advanced**->**Accounts** to configure the account settings.

Refer to **Configuring via Web Page** for the parameter details.

Overview

Keypad Instruction



You can check the following list which introduces the IP phone's keypad in details:

Power Indication LED

It will show the power status, it will be on if the phone is powered, off if the phone is not powered，and blink when someone calls in or there is a call on mute.

Soft Key

The screen will display labels for these keys, to identify their context-sensitive functions.

DTMF Keypad

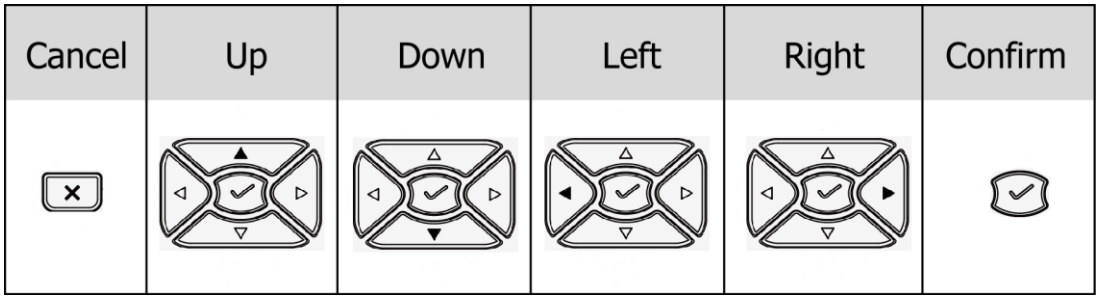
Use the DTMF hard keys to enter numbers, letters and special characters. Depending on the selected input mode, you can enter digits, lower / upper case or special characters.

DSS Keys

These keys are used for various functions such as Line, Speed dial, Intercom, BLF, URL, etc. The corresponding LED will light when the assigned function is turned on.

Navigation Keys

Press the navigation keys to navigate in the display menus, confirm or cancel actions.



Audio Device Control Keys

Use the audio device control keys to perform the following actions depending on your phone type:

: Adjust the volume of the handset, headset, speaker and ring tone;

: Allows for hands-free communication during calls;

: Place and receive calls through an optionally connected headset, the LED will be on when the phone is in Headset mode;

: Mute audio transmission locally during calls;

Hard Feature Keys

: Allow users to access the voicemail directly;

: Enable a setup of conference;

: Place a call on hold or resume it;

: During a call, press to transfer the current call to the third party;

: Press to enter the Dialed Calls interface, then select a record to call out.

LED Instruction

Table 1 DSS Keys set as BLF

LED Status	Description
Steady green	The monitored account is in idle status
Blinking red	There is an incoming call to the monitored account
Steady red	The monitored account is on a conversation
Off	It is not active as BLF

Table 2 DSS Keys set as BLA (Bridged Line Appearances)

LED Status	Description
Steady green	All of the members are in idle status
Steady red	Some part(s)is seizing the line
Blinking green 300ms	Some part(s) is ring-back
Blinking red 300ms	Some part(s) is ringing
Steady orange	Some part(s) is on the phone
Blinking Orange 500ms	Some part(s) is under the public hold status, and all of the members can retrieve the call
Blinking green 500ms	Some part(s) is under the private hold status, and only the initiator can retrieve the call
Blinking red 500ms	In three way conference, all of the parts press hold.
Off	It is not active as BLA

Table 2 Power Indication LED

LED Status	Description
Steady red	Power on
Blinking red	There is incoming call to the device, or there is a call on mute
Off	Power off

You can configure to disable the Power Indication LED via web user interface, click on Phone-> Features, and select **Enabled** from the pull-down list of **Close Power Light**.

Icon Instruction

The IP Phone displays different kinds of icons on its LCD. You can refer to the following table for their meanings:

Icon	Description
	Network is unavailable
	Hands-free speakerphone mode
	Handset mode
	Headset mode
123	Numeric input mode
abc	Multi-lingual lowercase letters input mode
ABC	Multi-lingual uppercase letters input mode
2aB	Alphanumeric input mode
	Voice Mail
AA	Auto Answer

Icon	Description
DND	Do Not Disturb
	Call Forward
	Call Mute
	Keypad Lock
	Received Calls
	Dialed Calls
	Missed Calls
	Recording box is full
	A call cannot be recorded
	Recording starts successfully
	Recording cannot be started
	Recording cannot be stopped

User Interface

There are two ways to customize specific options on your phones:

1. Using keypad and display on the phone.
2. Using web user interface in an Internet browser from your PC. Refer to **Configuration and Registration** to access the web user interface.

In many instances, it is possible to use both the user interfaces to operate the phone and change settings. However, are only possible via a phone or web user interface. Please refer to the following table for differences:

Phone Options	Phone UI	Web UI
Status --IP --MAC --Firmware --Network --Phone --Accounts	√	√

Phone Options	Phone UI	Web UI
Features		
--Call Forward	√	
--Call Waiting	√	
--DSS Keys	√	
--Key as Send	√	
--Hot Line	√	
--Anonymous Call	√	√
--Auto Redial Settings	√	
--Dial Plan		
--Tones		
--DND Code	√	
--Emergency		
--Intercom	√	
--Call Completion	√	
Basic Phone Functions		
--Language	√	√
--Time & Date	√	√
--Ring Tone	√	√
--Phone Volume	√	
--Logo Customization		√
Advanced Phone Functions		
--Accounts	√	
--Network	√	
--Keypad Lock	√	
--Reset to Factory	√	√
--Set Admin Password	√	
--Update		
--Set AES Key	√	
--Auto Provision	√	
SIP Account		
--User Option	√	
--Account Active	√	
--Label	√	
--Display Name	√	√
--Register Name	√	
--User Name	√	
--Password	√	
--SIP Server	√	
--Register Status	√	

Phone Options	Phone UI	Web UI
--Server Option		
--SIP Server	√	
--SIP Server Port		
--Outbound Status	√	
--Outbound Proxy Server	√	
--Backup Outbound Proxy Server		√
--NAT Traversal		
--STUN Status	√	
--STUN Server	√	
--Missed Call Log		
--Auto Answer	√	
Other Features		
--Messages		
--History	√	√
--Directory		√
--Local Directory		√
--Blacklist		√

Note:
1. The above table only indicates most of phone functions rather than all of them. Please refer to the relating parts for more information.

The default administrator password is **admin**.

Customizing Your Phone

General Settings

Phone Status

You can view the status of your phone using the phone user interface or the web user interface.
This option allows you to review:

- Network status: IP Address, Mac Address, Gateway, DNS, WAN, LAN, etc;
- Phone status: Model, Hardware version, Firmware version, Product ID, MAC, etc;
- Accounts: The 3 SIP accounts status.

To check the phone status via phone user interface:

1) Press directly to check the IP.
2) Press the navigation keys to check other information.

To view the phone status via web user interface:
Open the web browser and input the IP Address http://WAN-ip-address; Enter the username and password (default username and password for administrator are **admin/admin**), click **Status** to check the status.

Language

The default language of phone user interface is English. The default language of web user interface depends on your computer's operation system. It will automatically match the language with your computer and browser.
It supports Simplified Chinese, Traditional Chinese, French, German, Italian, Polish, Turkish, Portuguese, Spanish, etc. You can change the language of phone user interface and web user interface independently.

Note:
All languages may not be available for selection. The available languages depend on the language packs currently loaded to the IP phone. Contact your system administrator for more information about loading language packs.

To change the language of phone user interface via phone user interface:

1) Press ->**Settings**->**Basic**->**Languag**e.
2) Press the navigation keys to select the desired language.
3) Press to save the change or to cancel.

To change the language of web user interface via web user interface:

1) Click on **Phone**-> **Preference**.
2) Select the desired language from the pull-down list of **WEB Language**.
3) Click **Confirm** to save the change

Time and Date

The time and date appears on the idle screen of the IP phone. If the phone cannot obtain a time and date from the SIP server, please contact your system administrator if the time or date is incorrect. You can set the time manually or via the NTP server which is used to synchronize the time.

To configure time and date via phone user interface:

1) Press ->**Settings**->**Basic**->**Time & Date**.
2) If **SNTP** is selected, the phone will automatically get the time from the specific NTP Server. Press the navigation keys to select the specific options and do the relating changes. You can set the Time Zone, NTP Server1/Server2, and DST respectively.
3) If **Manual** is selected, the time can be set manually. Press the navigation keys to select the option and enter the specific date and time.
4) Press to save the change.
   The time appears on the idle screen will be changed.

To configure time format via phone user interface:

1) Press ->**Settings**->**Basic**->**Time & Date**->**Time & Date**.
2) Press the Left/Right navigation keys to select a preferred time format: 12 hour or 24 hour.
3) Press to save the change.

To configure daylight saving time via phone user interface:

1) Press ->**Settings**->**Basic** ->**Time & Date**->**SNTP**.
2) Scroll to **DST** field, and then press the navigation keys to select the desired option.
3) Press to save the change.

To configure daylight saving time via web user interface:

1) Click on **Phone**->**Preference**.
2) Select the desired option from the pull-down list of **Daylight Saving Time**, and enter other information required.
3) Click **Confirm** to save the change.

Keypad Lock

You can lock the keypad of your phone when you are temporarily not using it. This function helps you to protect your phone from unauthorized use. You can lock the following specific keys:

Menu Key: The key cannot be used until unlocked. You cannot access the menu of the phone.

Function Keys: The hard function keys cannot be used until unlocked. You cannot access the History, Message, Conference, Hold, Mute, Transfer, Redial, √, **X** keys, volume adjustment keys, navigation keys, DSS keys, and the soft keys.

All Keys: All of the keys cannot be used until unlocked. You can only use the phone to answer the incoming calls.

Lock&Answer: All of the keys except the key cannot be used until unlocked. The incoming calls will be answered automatically and cannot end the call by your party.

To enable keypad lock via phone user interface:

1) Press ->**Settings**->**Advanced**, enter the password, and then press .
2) Select to **Keypad Lock** option, and then press .
3) Press the navigation keys to select the desired keypad lock type.
4) Press to save the change or to cancel.
   If keypad lock is enabled, the icon will be displayed on the top right corner of the idle screen.
   If the keypad lock type is Lock&Answer, an additional icon AA will be displayed next to the icon.

To unlock the phone via phone user interface:

1) Press , you are prompted for the password.
2) Enter the password, and then press , the phone will be unlocked.
   The icon will disappear from the idle screen.
3) If the keypad lock type is Lock&Answer, you should press -> **Settings**-> **Advanced**->**Keypad Lock** and select **Disable** from the **Keypad Lock** field to unlock the keypad lock

To enable keypad lock via web user interface:
Click on **Phone**->**Preference** to do the relating changes. Please refer to the instructions above for the parameters' detail.

Note:
1. The default password for unlock is **admin**.
2. Users can make emergency calls when the phone is locked.

Audio Settings

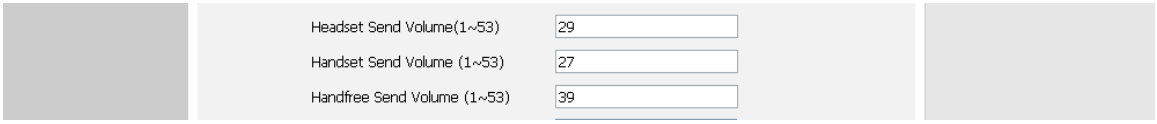
Volume

Sender Volume

You can adjust the sender volume of headset/handset/speakerphone via web user interface.

To adjust the sender volume via web user interface:

1) Click on **Phone**->**Features**.
2) Enter the desired volume in the **Headset Send Volume**, **Handset Send Volume** or **Handfree Send Volume** field.



3) Click **Confirm** to save the change.

Receiver Volume

You can adjust the receiver volume of handset/speakerphone/headset via phone interface.

To adjust the volume when you are not in an active call:

1) Press ->**Settings**->**Basic**-> **Phone Volume**.
2) Select **Handset Vol**, **Speaker Vol** or **Headset Vol**, and then press .
3) Press the volume adjustment keys or navigation keys to adjust the volume.
4) Press to save the change or to cancel.

To adjust the volume when you are in an active call:
When Handset/Headset/Speakerphone mode is activated, press the volume adjustment keys adjust the volume to a comfortable level.

Ringer Volume

You can adjust the ringer volume via phone interface.

To adjust the ringer volume via phone interface:

1) Press ->**Settings**->**Basic** -> **Phone Volume**.
2) Select the **Ringer Vol** and then press .
3) Press the volume adjustment keys or navigation keys to adjust the volume.
4) Press to save the change or to cancel.

You can also adjust the ringer volume by pressing the volume adjustment keys when the phone is idle.

Ring Tones

You can change the ring tone of your phone.

To adjust the ring tone of the phone via phone user interface:

1) Press ->**Settings**->**Basic**->**Ring tones**.
2) Press the navigation keys to select the desired ring tone.
3) Press to save the change or to cancel.

To change the ring tone of the phone via web user interface:

1) Click on **Phone**->**Preference**.

2) Select the desired ring tone from the pull-down list of **Ring Tones**.
3) Click **Confirm** to save the change.

Note:
You can delete a ring tone by selecting the desired one from the pull-down list of Ring Tones and clicking the Del key. The ring tone file of system cannot be deleted.

To specify ring tone for account via web user interface:
1) Click on **Account**->**Basic**.
2) Select the desired ring tone from the pull-down list of **Ring Tones**.
3) Click **Confirm** to save the change.

To upload a ring tone via web user interface:
1) Click on **Phone**->**Preference**.
2) Scroll to **Upload Ringtone** field**.**
3) Click **Browse** to select a ring tone file from your local system.
4) Click **Upload** to upload the file.
5) Click **Confirm** to save the change.

Note:
1. The ring tone file format must be in 16bits WAV format (via Ulaw Compression), 8K sample rate (monophony).
2. Blank or other special characters cannot be included in the file name.
3. The size of the ring tone file cannot exceed 100K.

Codec Selection

The IP phone supports the following audio codecs:
G722, G723-53, G723-63, G726-16, G726-24, G726-32, G726-40, G729, PCMU and PCMA.
You can enable or disable the desired codecs via web user interface. Contact your system administrator for more information about the codecs.

To enable/disable the codecs via web user interface:
1) Click on **Account**->**Codecs**.
2) Highlight the desired codec in the **Disable codecs** list or **Enable codecs** list, and click → or ← to move to the other list.
3) Click **Confirm** to save the change.

Note:
1. Codec Selection can only be configured via web user interface.
2. If codec G722 is negotiated, the LCD screen prompts call in process with HD voice.

Contact Management

Local Directory

You can store a large number of contacts in your phone's local directory. You can add, edit, delete, dial, or search for a contact in this directory.

To add a group to the local directory via phone user interface:

1) Press ->**Local Directory**.
2) Press Left/Right navigation key to select **AddGroup** and then press .
3) Enter the group name and select a ring tone.
4) Press to save the change or to cancel

To add a contact to the local directory via phone user interface:

1) Press ->**Local Directory**.
2) Press Up/Down navigation key to select the desired group.
3) Press Left/Right navigation key to select **Enter** and then press .
4) Press Up/Down navigation key to scroll to **<New Item> Add** and then press .
5) Enter the Name and the Office, Mobile or Other numbers.
   Press to change the input mode.
6) Press Left/Right navigation key to select an account from the **Account** field. If **Auto** is selected, the phone will use the first available account when placing calls to the contact from the local directory.
7) Press Left/Right navigation key to select a ring tone from the **Ring Tones** field.
8) Press to save the change or to cancel.

To edit a contact in the local directory via phone user interface:

1) Press ->**Local Directory**.
2) Press Up/Down navigation key to select the desired group.
3) Press Left/Right navigation key to select **Enter** and then press .
4) Press Up/Down navigation key to scroll to the desired contact
5) Press Left/Right navigation key to select **Edit** and then press .
6) Update the information of the contact.
7) Press to save the change or to cancel.

To delete a contact from the local directory via phone user interface:

1) Press ->**Local Directory**.
2) Press Up/Down navigation key to select the desired group.
3) Press Left/Right navigation key to select **Enter** and then press .
4) Press Up/Down navigation key to scroll to the desired contact.
5) Press Left/Right navigation key to select **Del** and then press .
   The LCD screen prompts "Delete Selected Item?".
6) Press to confirm the deleting or to cancel.

To search a contact via phone user interface:

1) Press ->**Local Directory**.
2) Press Up/Down navigation key to select the desired group.

3) Press Left/Right navigation key to select **Enter** and then press .
4) Enter the query condition using the keypad and the press .
   The qualified entry will be displayed on the LCD screen.

To add/delete/edit/move the contacts via web user interfaces:
Click on **Contacts**->**Local Directory**->**Contacts** to do the relating change. Please refer to the instruction above for the parameters' detail

To import contact list via web user interface:
1) Click on **Contacts**-> **Local Directory**->**Contacts**.
2) Click **Browse** to select a contact list file in .XML or .CSV format from your local system, and then click **Import XML** or **Import CSV**. The imported contacts will be showed in the local directory.

To export contact list, please refer to the instruction above.

Note:
Import/Export Contact List can only be operated via web user interface.

Blacklist

If you add a contact to blacklist, then the call from this contact cannot get through. Operating instructions of adding blacklists, editing blacklists, deleting blacklists, placing calls to blacklists and searching in the blacklists, refer to the operating instructions of **Local Directory**.

To add/delete/edit/move the contacts via web user interface:
Click on **Contacts**-> **Local Directory** ->**BlackList** to do the relating change. Please refer to the instruction above for the parameters' detail.

Other Settings

Key as Send

Users can set a specific key (or) to act as the send key.

To configure the send key via phone user interface:
1) Press ->**Features**->**Key as Send**.
2) Press Left/Right navigation key to select the desired send key: "**#**", "*****", or select **Disable** to disable this feature.
3) Press to save the change or to cancel.

To configure the send key via web user interface:
1) Click on **Phone**->**Features**.
2) Select the desired send key from the pull-down list of **Key As Send**.
3) Click **Confirm** to save the change.

You can enable or disable the Send Pound Key feature when the pound key is configured as send key. When the Send Pound Key feature is enabled, you can enter "##" without being dialed out. When the Send Pound Key feature is disabled, the dialed number will be dialed out immediately after pressing the pound key.

To enable/disable Send Pound Key feature via web user interface:

1) Click on **Phone**-> **Features**.
2) Select the desired send key from the pull-down list of **Send Pound Key**.
3) Click **Confirm** to save the change.

Hot Line

To configure hot line number via phone user interface:

1) Press ->**Features**->**Hot Line**.
2) Enter the **Hot Number** and the **HotLine Delay** (seconds).
3) Press to save the change.
   When you pick up the handset or press , it will dial out the number automatically if you do not press any key in the pre-set time specified in the **HotLine Delay** field.

To configure hot line number via web user interface:

1) Click on **Phone**-> **Features**.
2) Enter the hotline number and hotline delay in the **Hotline Number** and **Hotline Delay** fields.
3) Click **Confirm** to save the change.

Headset Prior

Headset prior allows the phone to use headset in priority.
To place a call using Headset Prior, enable the Headset Prior feature, physically connect your headset and press the Headset key to activate it for use. Press the desired numeric keys to place a call will now connect to the headset automatically.
To receive a call using Headset Prior, enable Headset Prior feature, physically connect your headset and press the Headset key to activate it for use, press the √ key to answer a call will now connect to the headset automatically.

To enable Headset Prior via web user interface:

1) Click on **Phone** -> **Features**.
2) Select **Enabled** from the pull-down list of **Headset Prior**.
3) Click **Confirm** to save the change.

Note:
The headset icon on the LCD screen or the Headset key in steady green indicates that the headset mode is activated.

Dual Headset

Dual headset allows users to use two headsets. To use this feature, you must connect your headsets to headset jack and handset jack respectively. Once the phone joins in a call, people with the headset connected to the headset jack has a full-duplex conversation, other people with the headset connected to the handset jack are only allowed to listen.

To enable Dual Headset via web user interface:

1) Click on **Phone** -> **Features**.
2) Select **Enabled** from the pull-down list of **Dual Headset**.
3) Click **Confirm** to save the change.

Anonymous Call

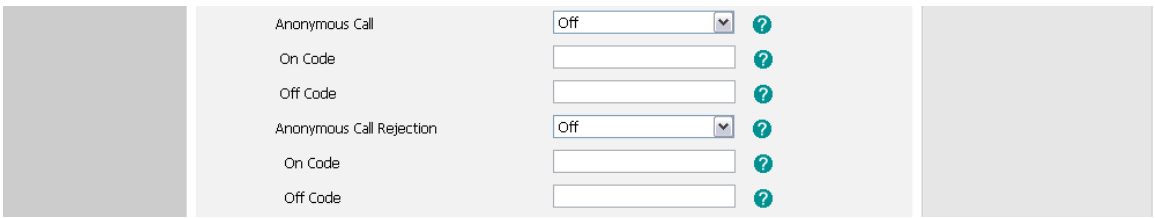
This feature allows you to make a call with the display of your calling identification information blocked. You can also configure to reject calls from callers who have blocked the display of their calling identification information (calling number and calling name).

To configure anonymous call via phone user interface:

1) Press ->**Features**->**Anonym Call**.
2) Press Left/Right navigation key to select an account from the **Line ID** field.
3) Press Left/Right navigation key to select **Enable**/**Disable** from the **Anonym Call** field.
4) (Optional.) Enter the anonymous call on code in the **Anonym On** field.
5) (Optional.) Enter the anonymous call off code in the **Anonym Off** field.
6) Press Left/Right navigation key to select **Enable**/**Disable** from the **Rejection** field.
7) (Optional.) Enter the anonymous call rejection on code in the **Rejection On** field.
8) (Optional.) Enter the anonymous call rejection off code in the **Rejection Off** field.
9) Press to save the change or to cancel.

Note:
This configuration is only available for the account selected in the Line ID field.

To configure the anonymous call via web user interface:
Click on **Account**-> **Basic** to do the relating change.



Please refer to the instruction above for the parameters' detail.

Auto Redial

Auto redial is a telephone feature that redials a busy number in a fixed number of times before giving up.

To enable Auto Redial via phone user interface:

1) Press ->**Features**->**Auto Redial**.
2) Press Left/Right navigation key to select **Enable** from the **Auto Redial** field.
3) Enter the interval in the **Interval** field.
4) Enter the number of times in the **Times** field.
5) Press to save the change or to cancel.

To configure auto redial via web user interface:

1) Click on **Phone**->**Features**->Auto Redial.
2) Select **Enabled** from the pull-down list of **Auto Redial**.
3) Enter the desired value in the **Auto Redial Interval** and **Auto Redial Times** fields.
4) Click **Confirm** to save the change.

Note:
If you enable the Auto Redial function, then without operations for 5 seconds in the auto redial interface, it will turn to the idle interface automatically.

Auto Answer

Auto-answer allows an incoming call to be answered without requiring any action by the user. This is a useful feature for people who have difficulty in using their hands or fingers, who have a visual impairment, or who have a cognitive impairment. You can set this function to a special account.

To enable Auto Answer via phone user interface:

1) Press ->**Settings**->**Advanced**->**Accounts**.
2) Press Up/Down navigation key to select an account and then press .
3) Press Up/Down navigation key to scroll to **Auto Answer** field.
4) Press Left/Right navigation key to select **Enable** from the **Auto Answer** field.
5) Press to save the change or to cancel.

To enable Auto Answer via web user interface:

1) Click on **Account**->**Basic**.
1) Select **Enabled** from the pull-down list of **Auto Answer**.
2) Click **Confirm** to save the change.

You can also configure the delay time before auto answer (seconds).

To configure Auto Answer Delay via web user interface:

1) Click on **Phone**->**Features**.
2) Enter the desired value in the **Auto Answer Delay** field.
3) Click **Confirm** to save the change.

Missed Call Log

You can configure whether to save the missed calls to the call history record or not. This feature can only be configured via web user interface:

1) Click on **Account**->**Basic**.
2) Select **Enabled** or **Disabled** from the pull-down list of **Missed Call Log**.
1) Click **Confirm** to save the change.

Logo Customization

You can configure a text logo which will be displayed on the idle screen.

To configure text logo via web user interface:

1) Click on **Phone**->**Features**.
2) Select **Enabled** from the pull-down list of **Use Logo**.
3) Enter a string in the **Text Logo** field.
4) Click **Confirm** to save the change.
5) Reboot the phone.
   Then the string will be displayed on the idle screen.

Programmable Keys

The navigation keys and function keys on the keypad are editable. Users can customize specific features for these keys according to their actual needs. (The programmable keys can only be available when the phone is idle.)
This function can only be set via web user interface:

1) Click on **DSS Keys**->**Programmable Keys**.
2) Customize specific features for the keys.
3) Click **Confirm** to save the change.

Live Dialpad

The feature defines whether to dial out the dialed number automatically.

To configure Live Dialpad via web user interface:

1) Click on **Phone**->**Preference**.
2) Select **Enabled** or **Disabled** from the pull-down list of **Live Dialpad**.
3) Click **Confirm** to save the change.

Replace Rule

A dial plan establishes the expected number. This includes country codes, access codes, area codes and all combinations of digits dialed. For example if you set the *Prefix* as 0 and *Replace* as 0086 (Chinese country code), when you dial 0 out, the number will be replaced by 0086 automatically.

To configure Replace Rule via web user interface:

1) Click on **Phone**->**Dial Plan**->**Replace Rule**.
2) To add a replace rule, enter the desired value in the **Prefix**, **Replace** and **Account** fields and then click **Add**.
3) To edit a replace rule, select the entry, update the information and then click **Edit**.
4) To delete replace rules, select the desired entries and then click **Del**.
5) Click **Save** to save the change.

Dial Now

Dial-now enables you to define the specific length of any number/letter in advance(for example xxx), next time when users dial 123 whose length matches the Dial-now rule, the phone will dial out 123 in one second without pressing the √ key.

To configure a Dial-now rule via web user interface:

1) Click on **Phone**->**Dial Plan**->**Dial now**.
2) To add a dial-now rule, enter the desired value in the **Dial-now Rule** and **Account** fields and then click **Add**.
3) To edit a dial-now rule, select the entry, update the information and then click **Edit**.
4) To delete dial-now rules, select the desired entries and then click **Del**.
2) Click **Save** to save the change.

You can also set time out for dial-now rule via web user interface. Click **Phone**-> **Features**, and then enter the time in the **Time Out for Dial-Now Rule** field.

Note:
If need to replace the unknown contents, then you can use (.) or (x), ".” stand for a string of char, “x” stand for any one char. The content in () stand for a variable, the first variable is expressed by $ 1, the second variable is expressed by $ 2, the rest can be done in the same manner. For example: if you want to replace the any input content with the content beginning with 8. Input (.) in Prefix box, and input 8$1 in Replace box.

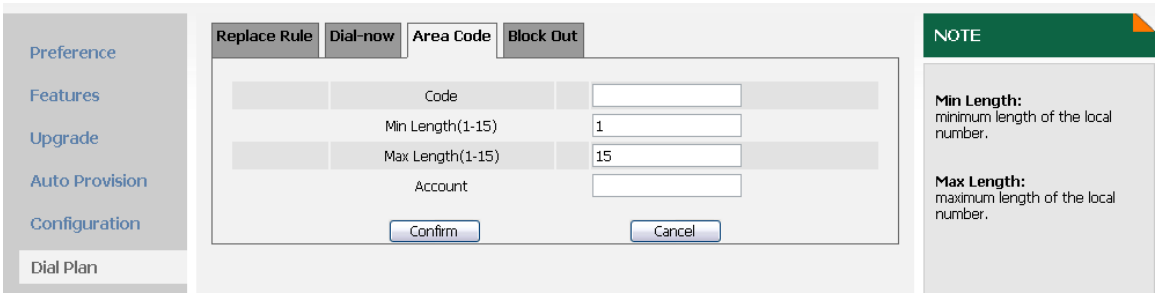
Area Code

Area codes are also known as Numbering Plan Areas (NPAs). These are necessary (for the most part) only when dialed from outside the code area and from mobile phones.

Area codes usually indicate geographical areas within one country, although the correlation to geographical area is becoming obsolete. For non-geographical numbers, as well as mobile telephones outside of the United States and Canada, the "area code" does not correlate to a particular geographic area.

To configure area code via web user interface:

1) Click on **Phone->Dial Plan->Area Code**.



2) Enter the **Code** and **Account**, set the **Min Length** and the **Max Length**.

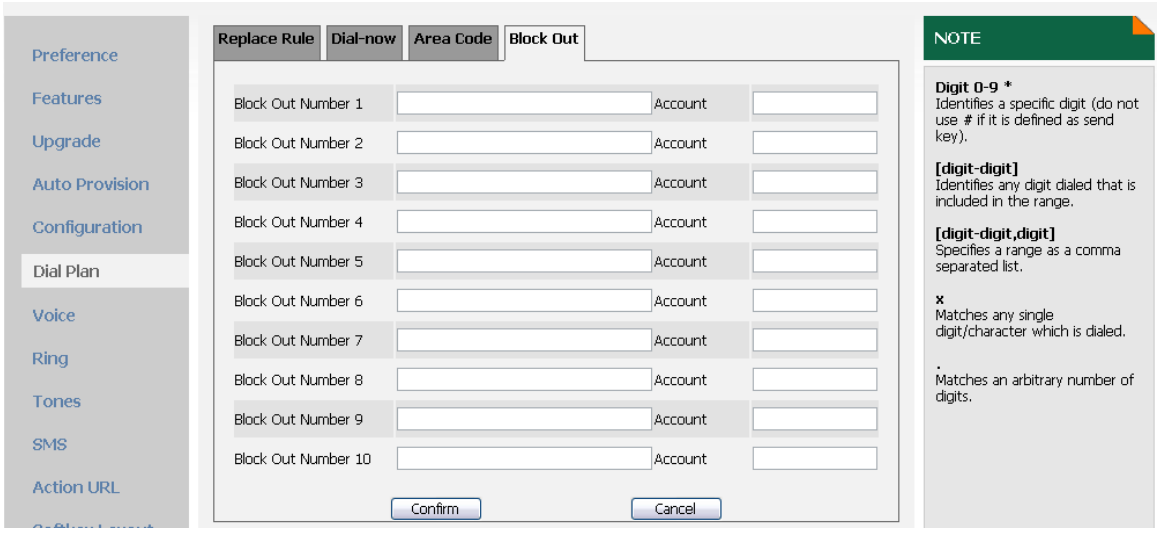
3) Click **Confirm** to save the change.

Block Out

The specific phone numbers can be forbidden to be call out from your IP phone.

To configure Block Out number via web user interface:

1) Click on **Phone->Dial Plan->Block Out**.



2) Enter the desired block out number and account in the proper fields.

3) Click **Confirm** to save the change.

Note:

1. The numbers set in Emergency cannot use the Dial Plan.
2. In the Account field, you can enter 1,2,3…, "1" represents Account 1, "2" represents Account 2 ……, if the account box is empty, it mean this rule works for all accounts .

Feature Synchronisation

When the synchronize feature is enabled, the change of DND or FWD feature will be synchronized between the device and the server.

To enable Feature Key Synchronisation via web user interface:

1) Click on **Phone**->**Features**.
2) Select **Enabled** from the pull-down list of **Feature Key Synchronisation**.
3) Click **Confirm** to save the change.

Push XML

Users configure the server's IP address on web page, when phone receives request, it will determine whether to display corresponding content on the phone sent by the specified server or not.

To configure Push XML via web user interface:

1) Click on **Phone**->**Features**.
2) Enter the IP address in the **PushXML Server IP** field.
3) Click **Confirm** to save the change.

WatchDog

When WatchDog feature is enabled, phone will auto reboot after 10 seconds if some important process of the phone crash. When the feature is disabled, the phone will not reboot.

To enable WatchDog via web user interface:

1) Click on **Phone**-> **Preference**.
2) Select **Enabled** from the pull-down list of **WatchDog**.
3) Click **Confirm** to save the change.

Using the Basic Call Functions

Making a call

Call Devices

You can make a phone call via the following devices:

1) Pick up the handset, icon will be shown on the idle screen.
2) Press , icon will be showed on the idle screen.
3) Press if the headset is connected to the phone in advance. The icon will be showed on the idle screen.

You can also dial the number first, and then select the method you will use to speak to the other party.

Call Methods

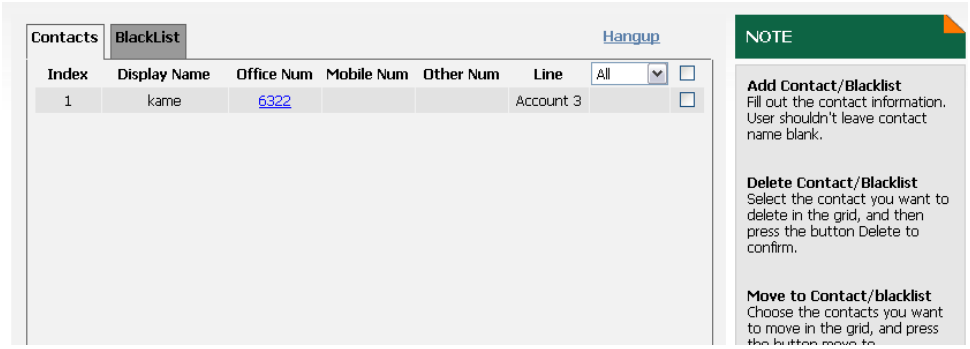
If you have registered more than one account, you can choose a certain account to make calls:

1) Press the Up/Down navigation key to select an account when your phone is idle (You should set the type of the Up and Down navigation keys to be **Switch Account Up** and **Switch Account Down** in advance, refer to **Programmable Keys**).
2) Press the DSS key which is configured as line key to active the corresponding account.

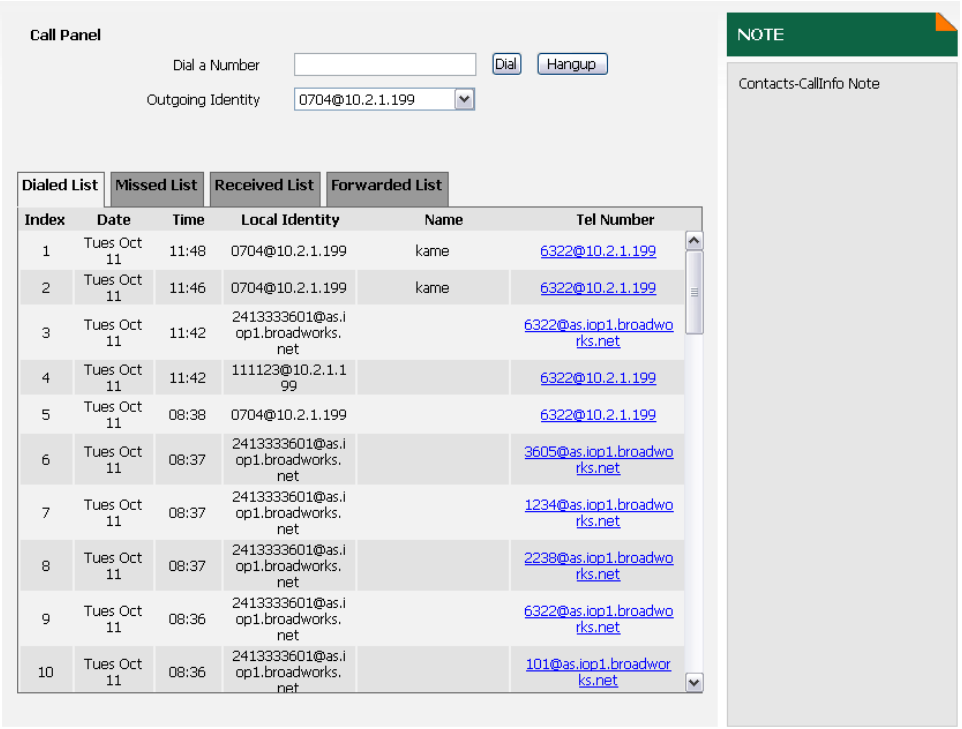
Then dial the number you want to call and then press or the pound key to call out.

You can also dial-up via web user interface:

1) Click on **Contacts**-> **Local Directory**->**Contacts/BlackList**.
2) Click the number which you want to dial out, and then the phone will dial out the number.



3) Or click on **Contacts**->**Phone Call Info**, enter the number in the **Dial a Number** field, select an account from the pull-down list of **Outgoing Identity**, and then click **Dial** to call out. You can click **Hangup** to end the call.

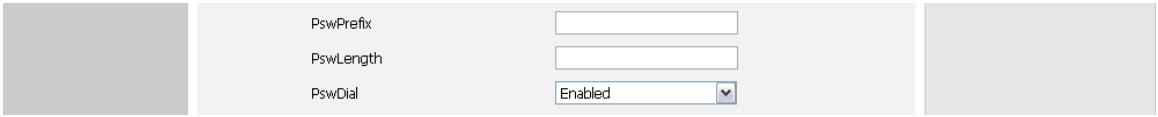

Call Panel
Dial a Number
Dial
Hangup
Outgoing Identity
0704@10.2.1.199
Dialed List
Missed List
Received List
Forwarded List
Index Date Time Local Identity Name Tel Number
1 Tues Oct 11 11:48 0704@10.2.1.199 kame 6322@10.2.1.199
2 Tues Oct 11 11:46 0704@10.2.1.199 kame 6322@10.2.1.199
3 Tues Oct 11 11:42 2413333601@as.iop1.broadworks.net 6322@as.iop1.broadworks.net
4 Tues Oct 11 11:42 111123@10.2.1.199 6322@10.2.1.199
5 Tues Oct 11 08:38 0704@10.2.1.199 6322@10.2.1.199
6 Tues Oct 11 08:37 2413333601@as.iop1.broadworks.net 3605@as.iop1.broadworks.net
7 Tues Oct 11 08:37 2413333601@as.iop1.broadworks.net 1234@as.iop1.broadworks.net
8 Tues Oct 11 08:37 2413333601@as.iop1.broadworks.net 2238@as.iop1.broadworks.net
9 Tues Oct 11 08:36 2413333601@as.iop1.broadworks.net 6322@as.iop1.broadworks.net
10 Tues Oct 11 08:36 2413333601@as.iop1.broadworks.net 101@as.iop1.broadworks.net
NOTE
Contacts-CallInfo Note

Password Dial

When number entered is beginning with the password prefix, the following N numbers after the password prefix will be hidden as *, N stand for the value which you enter in the PswLength field. For example: you set the password prefix is 3, enter the PswLength is 2, then you enter the number 34567, it will display 3**67 on the phone.

To enable Password Dial via web user interface:

1) Click on **Phone**->**Features**.
2) Select **Enabled** from the pull-down list of **PswDial**.



3) Enter the password prefix in the **PswPrefix** field
4) Enter the password length in the **PswLength** field.
5) Click **Confirm** to save the change

Call Completion

Have you encountered such a situation? When you call a contact, but the other side is busy on a call. Do you want the server to inform you immediately when the contact ends the call in order to establish a conversation in time? Call Completion can help you to solve this problem.

To enable Call Completion via phone user interface:

1) Press ▸☰ ->**Features**->**Call Completion**.
2) Press Left/Right navigation key to select **Enable** from the **Call Completion** field.
3) Press ✓ to save the change or ▸☰ to cancel.

Answering a Call

Answering an Incoming Call

1) If you are not on an active call, lift the handset to answer it using the handset, or press the Speaker key to answer it using the speakerphone, or press the Headset key to answer it using the headset.
2) If you are on an active call, the LCD will prompt: *New: xxx*. Press ✓ to answer the new call or × to reject the new call.

During the conversation, you can alternate between the Headset, Handset and Speakerphone modes by pressing the corresponding key or picking up the handset.

Denying an Incoming Call

Press × to deny the incoming call directly.

DND

Press **DND** to active DND Mode. Further incoming calls will be rejected and the LCD will display the DND icon. Press **DND** again to deactivate DND mode. You can find the incoming call record in the Call History.

To configure DND feature via phone user interface:

1) Press ▸☰ ->**Features**->**DND**.
2) Press Left/Right navigation key to select **Enable**/**Disable** from the **DND** field.
3) Enter DND on code in the **DND On Code** field.
4) Enter DND off code in the **DND Off Code** field.
5) Press ✓ to save the change or ▸☰ to cancel.

Call Forward

This feature allows you to forward an incoming call to another phone number, such as a cell phone or voice mailbox.

The following call forwarding events can be configured:

- *Always:* Incoming calls are immediately forwarded.
- *Busy:* Incoming calls are immediately forwarded when the phone is busy.
- *No Answer:* Incoming calls are forwarded when the phone is not answered after a specific ring times.

To enable Call Forward via phone user interface:

1) Press ▸≣ ->**Features**->**Forward**.
2) Select the desired forward type, and then press ✓.
3) Press Left/Right navigation key to select **Enable** from the proper field.
4) Enter the number in the **Forward to** field.
5) (Optional.) Enter On Code in the **On Code** field.
6) (Optional.) Enter Off Code in the **Off Code** field.
7) Press ✓ to save the change or ▸≣ to cancel..

To configure Call Forward via web user interface:
Click on **Phone**->**Features** to do the relating change. Please refer the configuration information above.

Intercom

Intercom mode is useful in an office environment as a quick access to connect to the operator or the secretary.

To configure Intercom via phone user interface:

1) Press ▸≣ ->**Features**->**Intercom**.
2) Press Left/Right navigation key to select **Enable**/**Disable** from the **Intercom Allow** field to set whether to answer the incoming intercom calls.
3) Press Left/Right navigation key to select **Enable**/**Disable** from the **Intercom Mute** field to set whether to mute the incoming intercom calls automatically.
4) Press Left/Right navigation key to select **Enable**/**Disable** from the **Intercom Tone** field to set whether to play ring tones when there is incoming intercom calls to your extension.
5) Press Left/Right navigation key to select **Enable**/**Disable** from the **Intercom Barge** field to set whether to answer the incoming intercom calls during a conversation. If the option is enabled, when there is an incoming intercom call to your extension and you are on an intercom conversation, it will refuse the call automatically. Otherwise it will put the current call on hold and answer the incoming intercom.
6) Press ✓ to save the change or ▸≣ to cancel.

During an Active Call

Mute

This function allows you to mute the microphone of the active audio device during a call. You cannot be heard by the other party, but still hear all other parties while mute is enabled. When you press the Mute key, all of the conversation will be muted.

To mute/resume a conversation:
Press during a conversation to mute the call, the icon will be shown on the LCD screen, and the power indication LED will blink. Press the key again to get the phone return to normal conversation.

You can configure to enable or disable the Mute feature via web user interface.

To enable/disable Mute feature via web user interface:
1) Click on **Phone**->**Features**.
2) Select **Enabled** or **Disabled** from the pull-down list of **Allow Mute**.
3) Click **Confirm** to save the change.

Call Hold

This call function allows you to place an active call on hold. In this case your IP PBX might play a melody or message to the other party while waiting. Other calls can be received and made while having a call on hold.

To hold/resume a call:
1) Press to put your active call on hold.
2) If there is only one call on hold, press again to retrieve the call.
3) If there is more than one call on hold, press the Up/Down navigation key to select a call, and then press to retrieve the call.

The phone will beep softly to remind you that you have calls on hold if Play Hold Tone feature is enabled.

To enable Play Hold Tone feature via web user interface:
1) Click on **Phone**->**Features**.
2) Select **Enabled** from the pull-down list of **Play Hold Tone**.
3) Enter the delay time in the **Play Hold Tone Delay** field.
4) Click **Confirm** to save the change.

Note:
When the call is placed on hold, put down the handset will change the conversation to the Speakerphone mode instead of ending the call.

Call Waiting

This call feature allows your phone to accept other incoming calls to the extension no matter under which circumstances.

To enable/disable Call Waiting via phone user interface:

1) Press ▸≡ ->**Features**->**Call Waiting**.
2) Press Left/Right navigation key to select **Enable**/**Disable** from the **Call Waiting** field.
3) Press Left/Right navigation key to select **Enable**/**Disable** from the **Play Tone** field. This option used to define whether to play ring tones when there is an incoming during an active call.
4) Press ✓ to save the change or ▸≡ to cancel.

To enable/disable Call Waiting via web user interface:
Click on **Phone**->**Features** to do the relating change.

Call Transfer

You can customize your phone so that incoming calls are transferred to the third party such as another extension, mobile phone number, etc. There are three ways to transfer the call: Blind Transfer, Attended Transfer and Semi-Attended Transfer.

To perform a Blind Transfer via phone user interface:

1) Press C-C during an active call, the call is placed on hold.
2) Dial the telephone number of the third party and then press C-C.
3) Press × to finish the transfer.

To perform a Semi-Attended Transfer via phone user interface:

1) Press C-C during an active call, the call is placed on hold.
2) Dial the telephone number of the third party and then press #SEND.
3) Press C-C when receiving ring-back tone.
4) Press × to finish the transfer.

To perform an Attended Transfer via phone user interface:

1) Press C-C during an active call, the call is placed on hold.
2) Dial the telephone number of the third party and then press #SEND.
3) Press C-C when C answers the call.
4) Press × to finish the transfer.

Note:
Make sure that the SIP server you have registered supports this function.

3-way Conference

You can establish a three-party conference, during the conversation three phone parties can communicate with each other.

To establish a conference:

1) Press [ ] during an active call, the call is placed on hold.
2) Dial the number of the third party and then press [#SEND] .
3) Press [ ] when the third party answers the call. Then the three parties are now included in the conference.
4) Press [ ] during the conference to place the conference on hold.
5) Hang up to disconnect all parties.

Network Conference

If you want to make a conference with more than three people, you can enable the function of network conference. This function needs the server's support.
If you enabled this function, you can put the conference on the server.

To enable Network Conference via web user interface:

1) Click on **Account** ->**Advance**.
2) Select an account from the pull-down list of **Account**
3) Select **Network** from the pull-down list of **Conference Type**.
4) Enter the conference URI in the **Conference URI** field.
5) Click **Confirm** to save the change.

To establish a conference:

1) Press [ ] during an active call, the call is placed on hold.
2) Dial the number you want to add into the conference, and then press [#SEND] .
3) When the call is answered, press [ ] .
4) Repeat the steps until you have added all intended parties into the conference.
5) Press [ ] during the conference to hold local call without influencing other parties in the conference.

Voice Mail

Your voice mailbox messages, which are usually stored on a media server of your local or hosted VoIP telephony system, can be accessed from your phone.
New voice messages can be indicated both acoustically and visually as described below:

- The idle screen will indicate there are new voice messages coming:
- The Message key LED will be lighted.

To configure the voice mail code via phone user interface:

1) Press [▸≡] ->**Messages** ->**Set Voice Mail**.
2) Press Left/Right navigation key to access the account which you want to set, enter

the code voice mail code in the proper field.

3) Press to save the change or to cancel.

Note:
Contact your system administrator for the connecting code. Different systems have different codes.

To retrieve the new voicemail via phone user interface:

1) Press .
2) You may be prompted to enter the password which is needed to connect to your VoIP telephony system. It depends on your system.
3) Your voice mailbox is called and you are able to listen to your new and old messages.

Note:
Before retrieving the new voicemail, please make sure that the connecting code has been set on the phone.

Using the Advanced Phone Functions

Account Setting

Refer to **Configuration and Registration** for the basic account setting information. The following table lists the instruction of the field about the advanced Account Setting.

Field Name	Description
UDP Keep-alive Message	Defines whether to active the phone UDP Keep-alive mechanism. The default value is Enabled.
UDP Keep-alive Interval	This parameter specifies how often the phone will send a packet to the SIP server. The default value is 30 seconds.
Login Expire	This parameter specifies the time frequency that phone refreshes its registration. The default value is 3600 seconds.
Local SIP Port	Local SIP port. The default value is 5060.
RPort	The parameter allows you configuring the proxy to send responses back to a particular address and port. The default value is Disabled.
SIP Session Timer	This document defines an extension to the Session Initiation Protocol (SIP). This extension allows for a periodic refresh of SIP sessions through a re-INVITE or UPDATE request. The refresh allows both user agents and proxies to determine if the SIP session is still active.
Subscribe Period	This parameter could set the period of the subscription. The default value is 3600 seconds.
DTMF Type	Select the DTMF type.

You can only configure these settings via web user interface.

1) Click on **Account**.
2) Select the desired account from the pull-down list of **Account**.
3) Click on **Advanced** to do the relating settings.

Contact your system administrator for more information.

TLS

TLS(Transport Layer Security), an IETF standards track protocol(RFC 5246), was based on the earlier SSL specifications developed by Netscape Corporation.
If you make a call based on TLS and SRTP (Set the transport as TLS and the SRTP as On), the IP phone UI will display the connecting and ring back interface. If the reception also set the transport as TLS, then IP phone UI will display the icon on both sides.

To configure TLS via web user interface:

1) Click on **Account**->**Basic**.
2) Select the desired account from the pull-down list of **Account**.
3) Select **TLS** from the pull-down list of **Transport**.
4) Click **Confirm** to save the change.

DNS-SRV

If the SIP server cannot be used, the phone will be connected on the server which is available.

To configure DNS-SRV via web user interface:

1) Click on **Account**->**Basic**.
2) Select the desired account from the pull-down list of **Account**.
3) Select **DNS-SRV** from the pull-down list of **Transport**.
4) Click **Confirm** to save the change.

Network Setting

PC Port Setting

Please refer to **Configuration and Registration** for the basic network WAN setting information. The following table lists the instructions of the field about the network PC port setting.

Field Name	Description
As Bridge	If you select the Bridge mode, then the two Fast Ethernet ports will be transparent.
As Router	If you select the Router mode, the SIP phone will work as a router.
IP address	User could configure the PC port IP address.
Enable DHCP Server	If you set the DHCP server on, the device connected to the PC port will get the IP address automatically between the start IP address and the end IP address. But if you select the bridge mode, the DHCP server cannot work.
Start IP Address	Indicate the range of the IP address.
End IP Address	Indicate the range of the IP address.

To configure PC Port settings via phone user interface:

1) Press ->**Settings**->**Advanced**.
2) Enter the password required, scroll to **Network** option, and then press .
3) Scroll to **PC port** and then press .
4) If you select **Bridge**, it will save and return to the previous menu.

5) If you select **Router**, enter the IP address, subnet mask in the proper fields, enable/disable DHCP server.
6) Press to save the change or to cancel.

To configure PC Port via web user interface:
Click on **Network**->**Basic**-> **LAN** to do the relating configuration. Contact your system administrator for more information.

VLAN Setting

VLAN is a group of hosts with a common set of requirements that communicate as if they were attached to the Broadcast domain, regardless of their physical location. The following table lists the instruction of the field about the VLAN Setting.

Field Name	Description
Voice QoS	When the network capacity is insufficient, QoS could provide priority to users by setting the value.
Local RTP Port	Define the port for voice transmission.
WebServer	Users can select the WebServer type: Disable, HTTP, HTTPS, or HTTPS & HTTP.

To configure VLAN settings via phone user interface:
1) Press ->**Settings**->**Advanced**.
2) Enter the password required, scroll to **Network** option, and then press .
3) Scroll to **VLAN** and then press .
4) Scroll to **WAN Port** and then press .
5) Set the **VLAN Status**, and enter the **VID Number**, **Priority**.
6) Press to save the change or to cancel.
7) Refer to the steps above to set the **PC Port** option.

To configure VLAN settings via Web user interface:
Click on **Network**->**Advanced**->**VLAN** to do the relating configuration. Contact your system administrator for more information.

LLDP

The Link Layer Discovery Protocol (LLDP) is a vendor-neutral Layer 2 protocol that allows a network device to advertise its identity and capabilities on the local network.

Enable LLDP function, the phone will go to switch to get related VLAN parameters automatically (Synchronous with VLAN in switch).

To configure LLDP settings via web user interface:
1) Click on **Network**->**Advanced**.

2) Scroll to **LLDP** field, and then select **Enabled** from the pull-down list of **Active**.

3) Enter the corresponding packet interval in the **Packet Interval** field.
4) Click **Confirm** to save the change

HTTPS

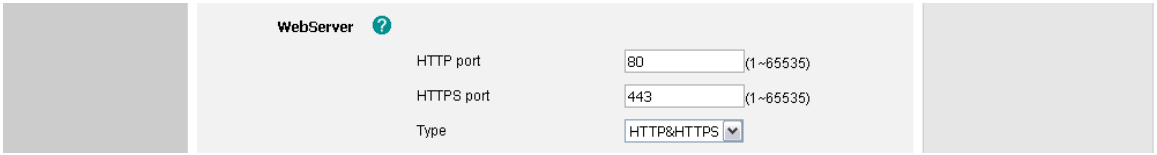
This IP phone can support HTTPS (Hypertext Transfer Protocol over Secure Socket Layer). Adding SSL layer under HTTP, in short, it is a security version of HTTP. Users can set this transmission mode via web page.

To configure HTTPS settings via phone user interface:

1) Press ->**Settings**->**Advanced**.
2) Enter the password required, scroll to **Network** and then press .
3) Scroll to **Web Type** and then press .
4) Press Left/Right navigation key to select the transmission mode.
5) Press save the change or to cancel.

To configure HTTPS settings via web user interface:

1) Click on **Network**->**Advanced**
2) Scroll to **WebServer** field, and then select the desired transmission mode from the pull-down list of **Type**.



3) Click **Confirm** to save the change.

Note:
1. Contact your system administrator for more information about the HTTPS.
2. IP phone also support Internet Protocol Version 6.

Maintenance Tasks

Administrator Mode

The phone allows two modes to configure the phone:

- User Mode
- Administrator Mode

Administrator mode grants unlimited access to the phone configuration on both Web and Phone interface. User Mode cannot set the following options: Advance settings of

accounts, Advance settings of Network, Reset to Factory, and other advance phone settings

Administrator/User Password

Administrator mode grants unlimited access to the phone configuration on both web and phone user interface. The administrator/user password is used to access:

- Web user interface.
- The advanced settings of the phone such as Account, Reset to Factory, and Advanced Settings via the phone interface.

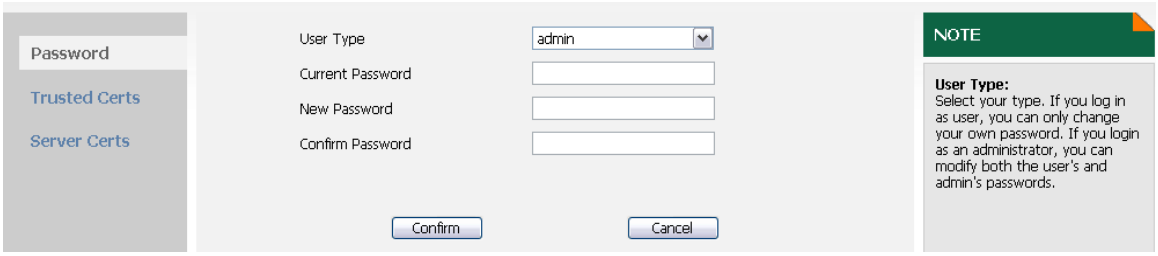
The default administrator password is **admin**. Meanwhile the user name for Web user interface access is **admin**.

To change the administrator password via phone user interface:

1) Press ->**Settings**->**Advanced**.
2) Enter the password required, scroll to **Set Password** and then press .
3) Enter the **Current PWD**, **New PWD** and **Confirm PWD**, press **abc** soft key to change the input method.
4) Press to save the change or to cancel.

To change the administrator password via web user interface:

1. Click on **Security**.
2. Select **admin** from the pull-down list of **User Type**, enter the **Current Password**, **New Password** and **Confirm password**.



3. Click **Confirm** to save the change.

To change the user password, refer to the steps introduced above.

Reboot

You should reboot the phone when you are challenged, e.g. after applying changes to the phone configuration.

To reboot the phone via web user interface:

1) Click on **Phone**->**Upgrade**.
2) Click **Reboot** to reboot the phone.

Note:
Please do not power off during reboot, or it will cause the flash memory error.

Reset to Factory

You should reset the phone only in this case: the phone configuration was changed and the phone is not functioning anymore. To maintain the configuration of the phone, you need your system administrator or service provider's advice.

To reset to factory via phone user interface:

1) Press ->**Settings**->**Advanced**.
2) Enter the password required, scroll to **Reset Factory** and then press .
3) You are prompted to confirm the change, press to reset to factory settings or to cancel.
   It takes a few minutes to reset the phone, please do not power off during the process, or it will cause flash memory error.

To reset to factory via web user interface:

1) Click on **Phone**->**Upgrade**.
2) Click **Reset To Factory**.

Note:

1. The reset to factory option must in the admin mode.
2. If you confirm to reset to factory, contact list, call history, account settings, etc will be lost. You need to export the configuration first if you still want to import the old configurations after reset. Or your phone must be configured manually unless mass provisioning is used!

Export/Import Configuration File

To Export/Import configuration file via web user interface:

1) Click on **Phone**->**Configuration**.
2) Scroll to **Export / Import Config** field.
3) Click **Export** to export the file to your local system.
4) Click **Browse** and select a configuration file on your local system, and then click **Import** to import the configuration file.
   It takes a few minutes to finish the process, please do not power off during the process, or it will cause flash memory error.

Firmware Update

The phone is delivered with pre-installed firmware. If you require updating the phone's firmware, please contact your system administrator. You can only update the firmware via web user interface.

To update the firmware manually via web user interface:

1) Click on **Phone**->**Upgrade**.
2) Scroll to **Select and Upgrade Firmware** field, and click **Browse** to select a firmware file on your local system.
3) Click **Upgrade** to update the firmware.
   Please do not power off during the updating process.

To update the firmware automatically via web user interface:

1) Click on **Phone**->**Auto Provision**.
2) Configure the relating settings: **PNP Active**, **DHCP Active**, **Custom Option**, **Server URL**, **Common AES Key**, **MAC-Oriented AES Key**, **Repeatedly**, **Interval**, etc.
3) Click **Confirm** to save the change. The phone will check the server for a new firmware in a specific time, and update automatically if there is a new firmware.
4) You can also update the firmware immediately by clicking **Autoprovision Now**.

To configure Auto Provision via phone user interface:

1) Press ▸≣ ->**Settings**->**Advanced** -> **Auto Provision**.
2) Enter the **URL**, **User Name** and **Password**.
5) Press ✓ to save the change or ▸≣ to cancel.

Decryption

This IP phone can support y000000000000.cfg and mac.cfg files encryption and decryption for user authentication to realize security usage. If there are any encrypted y000000000000.cfg or mac.cfg files on the server, uses can open the webpage of your IP phone.

To configure AES Key via web user interface

1) Click on **Phone**->**Auto Provision**.
2) Enter the values in the **Common AES Key** and **MAC-Oriented AES Key** fields.
3) Click **Confirm** to decryption the files and upgrade to the new version

Note:
Contact your system administrator for the decrypt password.

To configure AES Key via phone user interface:

1) Press ▸≣ ->**Settings**->**Advanced**.
2) Enter the password required, scroll to **Set AES Key** and then press ✓.
3) Enter the common and the MAC-oriented AES keys in the **Common** and **MAC-oriented** fields.
4) Press ✓ to save the change or ▸≣ to cancel.

Zero-sp-touch

Zero-sp-touch can help users to configure AUTOP and network parameters quickly. First to enable this function via web user interface, when the power is on or press the corresponding DSS key, the phone will turn to the zero-sp-touch interface.

To enable Zero-sp-touch when the power is on via web user interface:

1) Click on **Phone**->**Auto Provision**.
2) Select **Enabled** from the pull-down list of **Zero Active**.
3) Enter the time in the **Wait Time** field.
4) Click **Confirm** to save the change.

When reboot the phone, first a pull-down menu come into view.

1) No operation or press [×], it will return to idle status.
2) Press [✓] to enter Auto Provision setting interface. Enter the corresponding contents, and press [✓] to save the change or [▸≡] to cancel

System Log Export

If there are any errors happened in your phone, you can export the system log and send them to your system administrator for diagnosis.

To export the System Log via web user interface:

1) Click on **Phone**->**Configuration**.
2) Scroll to **Export System Log** field.
3) Select **Local** and then click **Export** to export the file to your local system.

You can also export the file to a server. Contact your system administrator for more information.

PCAP Trace Export

The PCAP Trace used to record the date transport of your IP phone. If there are any errors in your phone, you can export the PCAP trace and send them to your system administrator for diagnosis.

To export the PCAP Trace:

1) Click on **Phone**->**Upgrade**.
2) Scroll to **PCAP Trace** field, click **Start** to begin to capture the trace, and click **Stop** to stop capture the trace.
3) Click **Export** to export the file to your local system.

802.1X

IEEE 802.1X is an IEEE Standard for port-based Network Access Control (PNAC). It is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices attached to a LAN, either establishing a point-to-point connection or preventing it if authentication fails. It is used for securing wireless 802.1x access points and is based on the Extensible Authentication Protocol (EAP).
This IP phone can support 802.1X. Contact your system administrator for more information.

DSS Keys Configuration

The phone has 6 DSS keys which are able to set up to 29 functions per key.
The following list shows the functions you can set on the DSS keys and provides a description for each function. The default values of first two keys are Line and the default values of the last three keys are N/A which means the key hasn't been set for any functions.

- N/A
- Line
- Speed Dial
- BLF
- BLF List
- Voice Mail
- Pick Up
- Group Pickup
- Call Park
- Intercom
- DTMF
- Prefix
- Local Group
- Conference
- Forward
- Transfer
- Hold
- DND
- Redial
- Call Return
- Record
- URL Record
- Multicast Paging
- Group Listening
- Public Hold
- Private Hold
- Shared Line

- ACD
- Zero-sp-touch
- URL

Note:
1. Quick access functions like Intercom and Voice Mail must be pre-configured on your PBX in order to work on your phone. Contact your system administrator for more information.

Line

This key function is used to accept incoming calls and allows placing active calls on hold or resuming them. It can also be used to make a call using a specific extension

To configure the key as Line via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Line** from the pull-down list of **Type**.
4) Selected a line this functionality will be applied to from the pull-down list of **Line** When the value is **Auto**, press the DSS key in the idle state to make a call, the phone will select an available extension as the outgoing extension
5) Click **Confirm** to save the change.

Speed Dial

This key function allows you to speed up dialing numbers often used or hard to remember. Optionally, you can also configure a speed dial key to dial prefix numbers. With this option, the prefix numbers automatically dial when you press the DSS key.

To configure the key as Speed Dial via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Speed Dial** from the pull-down list of **Type**.
4) Enter the number you want to dial out in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.

BLF

This key function allows you to monitor the status (idle, ringing, or busy) of other SIP accounts. User can dial out by pressing a BLF DSS key.

To configure the key as BLF via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **BLF** from the pull-down list of **Type**.
4) Enter the number you want to monitor in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Enter the function codes in the **Extension** field.
7) Click **Confirm** to save the change.

Refer to **LED Instruction** for more information about the LED status in different situation.

Note:
In the web interface, you can enter pickup code in the **Extension** field to enable the pickup function. For example, if you enter 212 in the **Value** field, and enter the pickup code *83 in the **Extension** field, when there is an incoming call to 212, press the BLF DSS key, the call is picked up by the phone.

BLF List

This key function can monitor the list status, you can add accounts to the monitored list, the server sends messages to the phone to decide which account BLF list to be monitored.

To configure BLF List feature via web user interface:

1) Click on **Account**->**Advanced**.
2) Enter the BLF List URI in the **BLF List URI** field.
3) Enter the BLF List Code in the **BLF List Code** field.
4) Enter the BLF List Barge In Code in the **BLF List Barge In Code** field.
5) Click **Confirm** to save the change.

To configure BLF List key via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **BLF List** from the pull-down list of **Type**.
4) Selected a line this functionality will be applied to from the pull-down list of **Line**.
5) Click **Confirm** key to save the configuration.

Voice Mail

This key function is configured as Voicemail, which allows you to retrieve voicemail quickly by pressing this key.

To configure the key as Voice Mail via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Voice Mail** from the pull-down list of **Type**.
4) Enter the number you want to set as the voice mail box in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.

Pick Up

This key function allows you to specify extension that you want to monitor. when the monitored extension receives a call, you can press this DSS key to pick up the incoming calls.

To configure the key as Pick Up via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Pick Up** from the pull-down list of **Type**.
4) Enter the feature code (for example, input *78345, *78 is the feature code and the 345 is the extension number you want to pickup) in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.

Group Pick Up

This key function allows you to specify a group you want to monitor. You can add accounts to the monitored group, when the monitored group receives a call, you can press this key to pick up the incoming call. If the group receives multiple calls simultaneously, you will pick up the specific one the server assigns to you.

To configure the key as Group Pick Up via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Group Pick Up** from the pull-down list of **Type**.
4) Enter the feature code (for example,*79) in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.

Call Park

This key function allows you to place a call on hold at one phone and then retrieve the parked call from any other phone. When the conversation which is monitored is transferred to an unused extension, you can press this key to retrieve the call.

To configure the key as Call Park via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Call Park** from the pull-down list of **Type**.
4) Enter the feature code (for example, enter *681234, *68 is the call park code and the 1234 is the extension you want to park) in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.
7) Retrieve the parked call by dialing the call park retrieval code and the parked extension from any other phone.

Intercom

This key function is useful in an office environment as a quick access to connect the operator or the secretary. Before configure the Intercom DSS key, enable Intercom Allow via web user interface. Refer to **Intercom** for more information.

To configure the key as Intercom via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Intercom** from the pull-down list of **Type**.
4) Enter the extension number you want to intercom in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.

Note:
Intercom feature is not available on all servers. Contact your system administrator for more information.

DTMF

This key function allows you to send out the desired DTMF number during the conversation. The number needs to be set in advance.

To configure the key as DTMF via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **DTMF** from the pull-down list of **Type**.

4) Enter the specific number in the **Value** field.
5) Click **Confirm** to save the change.

Prefix

This key function allows you to dial a call with the prefix number. When press the DSS key in the idle state, the phone will be ready to make a new call, and show up the prefix number which you've set previously on the dial interface. You can enter other digit and then dial out

To configure the key as Prefix via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Prefix** from the pull-down list of **Type**.
4) Enter the specific number in the **Value** field.
5) Click **Confirm** to save the change

Local Group

This key function allows you to access Local Group quickly. Press the DSS key in the idle state, then you can enter the Local Group interface.

To configure the key as Local Group via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Local Group** from the pull-down list of **Type**.
4) Select a group from the pull-down list of **Line**.
5) Click **Confirm** to save the change.

Conference

This key function allows you to conference another party during a conversation.

To configure the key as Conference via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Conference** from the pull-down list of **Type**.
4) Enter the number you want to conference in the **Value** field.
5) Click **Confirm** to save the change.

You can press the DSS key in the idle state to speed dial the number specified in the **Vaule** field.

Forward

This key function allows you to forward a call to other account. Press the DSS key in the idle state, the phone will turn to the Always forward configure interface, then you can configure the Forward to number, when there is any call to the extension will be forwarded to the configure number automatically.

To configure the key as Forward via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Forward** from the pull-down list of **Type**.
4) Enter the extension you want to forward to in the **Value** field.
   If you leave the **Value** field blank, the DSS key acts as a Forward key.
5) Click **Confirm** to save the change.

Transfer

This key function allows you to perform the Blind/Attended/Semi-Attended Transfer during a conversation.

To configure the key as Transfer via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Transfer** from the pull-down list of **Type**.
4) Enter the extension you want to forward to in the **Value** field.
   During a conversation, press this key, the phone will blind transfer the call.
   If you leave the **Value** field blank, the DSS key acts as a Transfer key.
5) Click **Confirm** to save the change.

You can forbid the DSS key from performing Blind Transfer via web user interface by Clicking **Phone**->**Features**, and select **Disabled** from the pull-down list of **DsskeyBlindTran**.

Hold

This key function allows you to hold or retrieve a call during the conversation.

To configure the key as Hold via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
2) Select **Hold** from the pull-down list of **Type**.
3) Click **Confirm** to save the change.

DND

This key function allows you to activate the DND function immediately when you press the DSS key. Press it again to deactivate DND mode.

To configure the key as DND via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **DND** from the pull-down list of **Type**.
4) Click **Confirm** to save the change.

Redial

This key function allows you to access the **Dialed Calls** interface when press the DSS key in the idle state, then you can select a call to dial out or add it to the contacts, etc.

To configure the key as Redial via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Redial** from the pull-down list of **Type**.
4) Click **Confirm** to save the change.

Call Return

This key function allows you to call return the latest call you received.

To configure the key as Call Return via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select **Call Return** from the pull-down list of **Type**.
3) Click **Confirm** to save the change.

Record

This key function allows you to record during the conversation. Using this feature, please pay attention to the maximum record time and frequency in advance. Generally, it's only a few minutes.

To configure the key as Record via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Record** from the pull-down list of **Type**.
4) Click **Confirm** to save the change.

URL Record

This key function allows you to achieve the call recording capability during the conversation when follows the voice prompts.

1) During a conversation, press the DSS key to start recording.
2) Enable the Recording function, the recording icon will be flashing on the LCD.
3) Press the DSS key again to disable the recording function, the flashing recording icon will be disappeared simultaneously.
4) Follow the voice prompts to listen to the recording.

To configure the key as URL Record via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **URL Record** from the pull-down list of **Type**.
4) Enter the server URL in the **Value** field.
5) Click **Confirm** to save the change.

Note:
During a conversation, press this type of DSS key to start the recording process. If the other party hangs up, the phone will turn to idle state.

Multicast Paging

Multicast is the delivery of a message or information to a group of destination simultaneously in a single transmission from the source creating copies automatically in other network elements.
A multicast paging is essentially a predefined broadcast address that the phone is programmed to listen to. Each phone can be configured to listen to as many as 10 different multicast IP addresses. The priority of the IP Addresses is equivalent to the stream priority, 1 is the highest, 10 is the lowest. Streams with higher priorities will overlap those with lower priorities.
Users can configure the phone to send (receive) a Real Time Transport Protocol (RTP) stream to (from) pre-configured multicast address without involving SIP signaling.

To configure a DSS key for sending Multicast RTP stream via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Multicast Paging** from the pull-down list of **Type**.
4) Enter the multicast address (IP: port) in the **Value** field.
5) Click **Confirm** to save the change.

Note:
The multicast address ranges from 224.0.0.0 to 239.255.255.255.

Users can also configure the phone to use a default codec for sending Multicast RTP stream via web user interface:

1) Click on **Phone** -> **Features**.
2) Select a codec (i.e., **PCMU**) from the pull-down list of **MulticastCodec**.
3) Click **Confirm** to save the configuration.

To configure Listening Address(es) for receiving Multicast RTP stream via web user interface:

1) Click on **Contacts**->**MulticastIP**.
2) Enter the listening address which is preconfigured as multicast address (IP: port) in the **Listening Address** field.
3) Enter the label (String) which will appear on the LCD screen when receiving RTP from the corresponding Listening Address in the **Label** field.
4) Click **Confirm** to save the configuration.

After pressing a Multicast Paging key on the phone, the Phone sends RTP to a preconfigured multicast address(es) (IP:port). Any phone in the local network then listens for RTP on the preconfigured multicast address(es) (IP:port). For both sending and receiving of the multicast RTP there is no sip signaling involved. The phone receiving RTP will display the preconfigured listening multicast label (Label) to the user.

Note:
Multicast RTP is one way only- from sender to the multicast address(es) (receiver). For outgoing RTP multicasts, all other existing calls on the phone will be put on hold.

Group Listening

This key function allows you to enable the Speakerphone and Handset/Headset mode at the same time. It is suitable for the group conversation which has more than one person at one side. You are able to speak and listen by using handset/headset. Meanwhile the others nearby can listen by using speakerphone. You can get back to the previous mode by pressing the key again. (If the current mode is handset or headset, users can press the speaker key to open or close the group listening function)

To configure the key as Group Listening via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Group Listening** from the pull-down list of **Type**.
4) Click **Confirm** to save the change.

Public Hold

This key function allows particular BLA group to hold or retrieve a call during a conversation

To configure the key as Public Hold via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Public Hold** from the pull-down list of **Type**.
4) Click **Confirm** to save the change.

Private Hold

This key function allows all members belonging to that particular BLA group to hold the call, but only the initiator can retrieve the call during a conversation.

To configure the key as Private Hold via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Private Hold** from the pull-down list of **Type**.
4) Click **Confirm** to save the change.

Shared Line

Shared Line Appearances (SLA, which is also named as BLA) feature allows subscribers to share SIP lines and also provides status monitoring of the shared line. When a user places an outgoing call using such an appearance, all members belonging to that particular SLA group are notified of this status and are blocked from using this line appearance until the line goes back to idle or the call is placed on hold.
Similarly all members of the SLA group are notified of an incoming call and the call can be picked up on a line appearance associated with the SLA extension.

To configure the key as Shared Line via web user interface:
1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Shared Line** from the pull-down list of **Type**.
4) Enter the primary account number in the **Value** field.
5) Selected a line this functionality will be applied to from the pull-down list of **Line**.
6) Click **Confirm** to save the change.
   Please refer to **LED Instruction** for more information about the LED status in different situation.

Automatic Call Distribution (ACD)

The Phone supports an Automatic Call Distribution (ACD) feature for specific servers. The ACD feature allows the ACD system to distribute calls from a queue to registered IP phone users (agents). To use the ACD feature on the phone, the user should pre-configure a DSS key as ACD key.

To configure an ACD key via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **ACD** from the pull-down list of **Type**.
4) Selected a line this functionality will be applied to from the pull-down list of **Line**.
5) Click **Confirm** to save the configuration.

Once the user wants to subscribe to a queue (in order to receive incoming calls), the user presses the ACD key. The LCD screen prompts the user to specify the following information:

- **User ID**: the phone number(s) used to login into the queue.
- **Password**: the password used to login to the queue.

When ready to receive calls from the server, the phone user logs into a queue and changes the phone status to "available" manually using the IP phone UI (press the DSS Key again to enter to the ACD Status page, then select the phone status as **Available**/ **Unavailable**). If the phone is set to **Available**, then the server begins to distribute calls to this phone.

Note:
ACD feature is not available on all servers. Contact your system administrator for more information.

Zero-sp-touch

You can also press the DSS key which set as the Zero-sp-touch. Then press the DSS key, the phone will turn to the Zero-sp-touch interface.

To assign the key as Zero-sp-touch via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **Zero-sp-touch** from the pull-down list of **Type**.
4) Click **Confirm** to save the change

URL

If the key is configured as URL, then pressing this key, you can send HTTP requests to a web server.

To assign the key as URL via web user interface:

1) Click on **DSS Keys**->**Memory Keys**.
2) Select the desired DSS key.
3) Select **URL** from the pull-down list of **Type**.
4) Enter the number you want to set as URL in the **Value** field.
5) Click **Confirm** to save the change.

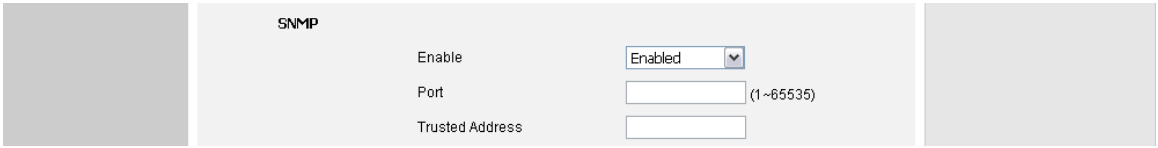
SNMP

Simple Network Management Protocol (SNMP) is an Internet-standard protocol for managing devices in IP networks. It is used mostly in network management systems to monitor network-attached devices for conditions that warrant administrative attention. SNMP exposes management data in the form of variables on the managed systems, which describe the system configuration. These variables can then be queried (and sometimes set) by managing applications. For further information, you can refer to http://en.wikipedia.org/wiki/SNMP.

By default, the phone does not accept SNMP requests. This is necessary because SNMP might introduce unwanted security threats. The phone only supports the GET request of SNMP. SET or other requests are not supported. To enable SNMP, you must specify what IP addressed may send SNMP requests to the phone.

To configure SNMP via web user interface:

1) Click on **Network** -> **Advanced**.
2) Scroll to **SNMP** field and select **Enabled** from the pull-down list of **Enable**.



3) Enter the port and trusted address in the **Port** and **Trusted Address** fields.
4) Click **Confirm** to save the change.

Note:

1. You may use several trusted IP addresses separated by space.
   For example, "192.168.3.10 192.168.3.78" will allow accesses from the two listed IP addresses.
2. Wildcards are not supported here.

The table below shows part of the available object identifiers (OID) on the phone:

OID	Description	MAX-ACCESS
1.3.6.1.2.1.37459.2.1.1.0	The textual identification of the contact person for this managed node, together with information on how to contact this person. If no contact	read-only
1.3.6.1.2.1.37459.2.1.2.0	An administratively-assigned name for this managed node. By convention, this is the node's fully-qualified domain name. If the name is unknown, the value is the zero-length string.	read-only
1.3.6.1.2.1.37459.2.1.3.0	The physical location of this node (e.g., `telephone closet, 3rd floor'). If the location is unknown, the value is the zero-length string.	read-only
1.3.6.1.2.1.37459.2.1.4.0	The time (in hundredths of a second) since the network management portion of the system was last re-initialized.	read-only
1.3.6.1.2.1.37459.2.1.5.0	The IP Phone's Firmware version	read-only
1.3.6.1.2.1.37459.2.1.6.0	The IP Phone's Hardware version	read-only
1.3.6.1.2.1.37459.2.1.7.0	Model of machine	read-only
1.3.6.1.2.1.37459.2.1.8.0	The IP Phone's MacAddress	read-only
1.3.6.1.2.1.37459.2.1.9.0	The IP Phone's IPAddress	read-only
1.3.6.1.2.1.37459.2.1.10.0	The IP Phone's LastUpVersion	read-only

We recommend a SNMP test suite which is available at http://www.net-snmp.org/.

The phone supports only SNMP v2c. Basic GET requests can be run in order to retrieve information from an SNMP-capable device. For example:

```
>snmpget v2c –c public 192.168.3.10 1.3.6.1.2.1.37459.2.1.5.0
```

Action URL

Action URLs are HTTP GET Requests allowing the phone to interact with web server applications. To use this feature, you should specify a HTTP URL corresponding to the predefined event. In case the predefined event has taken place, a HTTP GET to the specified URL is performed.

HTTP GET requests may contain variable names and variable values, which are separated by "=". Each variable value starts with $ in the query part of the URL (e.g. http://192.168.10.1/help.xml? MAC=$mac).
Action URLs can be triggered only by predefined events (e.g. Log on).

To configure Action URL via web user interface:

1) Click on **Phone**->**Action URL**.
2) Enter a HTTP URL in the proper field.
3) Click **Confirm** to save the change.

Note:
Action URL is allowed to be HTTPS URL. Contact with your system administrator for more information.

The following table provides variables can be used in the Action URL:

Parameter	Descriptions
$mac	MAC Address of the phone
$ip	IP Address of the phone
$model	Device model name of the phone
$firmware	Firmware version of the phone
$active_url	SIP URI of the active user
$active_user	User part of the SIP URI of the active user
$active_host	Host part of the SIP URI of the active user
$local	SIP URI of the caller
$remote	SIP URI of the callee
$display_local	Display name of the caller
$display_remote	Display name of the callee
$call_id	Call id
$duration	Call duration
$callDirection	Indication of the call as sender or recipient

Action URI

Opposite to Action URL, Action URI allows the phone to interact with web server applications by receiving and handling HTTP GET requests. To simply verify this feature, you can enter the following string into the browser address bar and press Enter:
http://192.168.10.5/cgi-bin/ConfigManApp.com?key=DNDOn
Then the phone with the IP Address of 192.168.10.5, if existed, will turn on DND.
Be aware that only predefined phone events are supported by the phone.

The following table provides the predefined phone events:

Phone Events	Descriptions
key=OK	Press the √ key
key=SPEAKER	Press the Speaker key
key=F_TRANSFER	Press the Transfer key
key=VOLUME_UP	Increase the volume
key=VOLUME_DOWN	Reduce the volume

Phone Events	Descriptions
key=MUTE	Press the Mute key
key=F_HOLD	Hold/Resume the call
key=X	Press the **X** key
key=0-9/*/POUND	Enter the DTMF number(include Numeric , * or # keys)
key=D1-D10	Press a DSS key
key=F_CONFERENCE	Press the Conference key
key=F1-F4	Press a soft key
key=MSG	Press the Message key
key=HEADSET	Press the Headset key
key=RD	Press the Redial key
key=UP/ DOWN/ LEFT/ RIGHT	Press the navigation key
key=Reboot	Reboot the phone
key=AutoP	Check the Auto provision
key=DNDOn	Enable DND
key=DNDOff	Disable DND
key=ATrans=XXXX	Attended Transfer to extension XXXX
key=BTrans=XXXX	Blind Transfer to extension XXXX
key=CALLEND	End a call

Music on Hold

Music on hold (MOH) is the business practice of playing recorded music to fill the silence that would be heard by telephone callers who have been placed on hold. To use this feature, you should specify a SIP URI pointing to the Music on Hold Server account. When a call is put on hold, the phone will invite this SIP URI to call the held phone to play music on hold. The Music on Hold Server is an application acting as a SIP client which automatically answers to SIP INVITE messages and immediately plays audio from any source located anywhere (LAN, Internet).

To configure Music on Hold via web user interface:

1) Click on **Account**->**Advanced**.
2) Enter the value of **Music on hold server**.

3) Click **Confirm** to save the change.

Note:

1. Syntax of the Music on hold server is the same as which of SIP URI.
2. All involved parties can not use encrypted RTP.

SIP Diversion Display

When an outgoing call from the phone is being diverted to another party (i.e., via call forward), the phone will display the Caller ID of the new party, Similarly, the new party will display the Caller ID of the original caller and the redirected call information indicated by DIVERSION or HISTORY-INFO header.

Call Diversion example:

1) Alice calls John.
2) John's phone is busy.
3) John's phone forwards the incoming call to another party (Mary).
4) Alice's phone displays the Caller ID of the new party(Mary)
5) Mary's phone receives a call and displays the caller ID of the initiator (Alice) and the diverted party (John).

Note:

1. Diversion is not available on all servers. Contact your system administrator for more information.
2. Redirected call information is sent by SIP message in the Diversion header, per draft-levy-sip-diversion-08, or the HISTORY-INFO header, per RFC 4244. It depends on the server.
3. Multiple diversions may take place on the phone. The phone receiving a diverted call (i.e., Mary's phone in example above) with the Diversion header will display the information of the last diverted party (last encountered). the phone receiving a diverted call (i.e., Mary's phone in example above) with the History-Info header will display the information of the first diverted party (first encountered).

Suppress DTMF Display

This feature allows the suppression of DTMF display when a number is dialed during an active call. When the Suppress DTMF Display is disabled, and you press number keys during an active call, the IP phone displays each digit as dialed in the LCD interface. When the Suppress DTMF Display is enabled, the IP phone will display "*" instead of dialed digits immediately in the LCD interface during the call.

To configure Suppress DTMF Display via web user interface:

1) Click on **Phone**->**Features**.
2) Select **Enabled** from the pull-down list of **Suppress DMTF Display**.
3) Click **Confirm** to save the change.

Tone Settings

You can use the country tone, or if you don't want to use the default one, you can custom it by yourself.
You can define the frequency and time period of all the following tones：

- Dial
- Ring Back
- Busy
- Congestion
- Call Waiting
- Dial Recall
- Record
- Info
- Stutter
- Message
- Auto Answer

To edit the tone filed via web user interface:

1) Click on **Phone**->**Tones**.
2) Enter the frequency and time period(in ms) as the following format: Frequency /Time Period (for example 400/200).
3) Press **Confirm** to save the change.

Note:

1. Contact your system administrator for more information about the frequency and time period parameters. You can enter up to 8 groups for each tone.
2. If the frequency is set as 0, it means silence.

Voice

To edit the Voice filed via web user interface:

1）Click on **Phone**->**Voice**.

2) Set the following parameters shown in the table.

Parameter	Description
Echo canceller	Defines whether to enable the echo canceller.
VAD	Voice activity detection (VAD), also known as speech activity detection or speech detection, is a technique used in speech processing in which the presence or absence of human speech is detected.
CNG	A comfort noise generator (CNG) is a program used to generate background noise for voice communications during periods of silence that occur during the course of conversation.

Parameter	Description
JITTER BUFFER	It is a shared data area where voice packets can be collected, stored, and sent to the voice processor in evenly.
Type	To select the type of JITTER BUFFER, adaptive or Fixed.
Delay	To set the Min Delay, Max Delay and Normal Delay parameter.

3） Click **Confirm** to save the change.

Ring

You can group your contacts, and then set the ringing tone for each group.

To edit the Ring option via web user interface:

1) Click on **Phone**->**Ring**.
2) Enter the group name in the **Internal Ringer Text** field.
3) Select a special ring tone from the pull-down list of **Internal Ringer File**.
4) Click **Confirm** to save the change.

Trouble Shooting

I cannot register to the server?

1) Check the IP address. If you set your WAN port in DHCP mode, please make sure that your DHCP server is on.
2) Check your gateway.
3) Check your DNS server.
4) Make sure your account information is the same as you have got from your ITSP.
5) Check whether the SIP server is on.
6) Check the SIP register port, the default value is 5060.

I can't get the IP address?

1) Make sure you have plugged the Ethernet cable into the WAN port.
2) Make sure that the DHCP server is on, and there are available IP addresses in the server.
3) Try to set your WAN port to static IP client mode.

During a call, I cannot hear any voice?

1) Make sure your handset is tightly connected with the phone.
2) Check whether you have muted the conversation or not.
3) Consult the outbound server details with your ITSP.

Have DTMF problem?

1) Check which kind of DTMF you are using, and whether it is compatible with the server.
2) Consult the payload value with your ITSP.

How to change the time?

Select the time zone or enter the time information manually on the webpage or the phone.

How to answer the incoming calls during a call?

If a call comes in when you are in a conversation, press the √ key to answer the incoming call.

How to refuse incoming calls during a call?

You can turn off the function of call waiting, and then our phone will refuse all the incoming calls when you are in a conversation.

How to update the firmware?

1) Enter the webpage of your phone, Click **Phone**->**Upgrade**.
2) Scroll to **Select and Upgrade Firmware** field, and then select a file from your local system.

3) Click **Upgrade** to update the firmware.

Note:
Make sure the firmware you select is provided by your service provider, or the device will probably crash after the update.

How to auto provision?
Consult the auto provision server address with your ITSP.

The manual is only for reference; please take the object as the standard.
We reserve the right to improve or change the product and the user guide without notice.